SPECIFICATION

TITLE OF THE INVENTION

VOICE CODE CONVERSION APPARATUS

BACKGROUND OF THE INVENTION

5   This invention relates to a voice code conversion apparatus and, more particularly, to a voice code conversion apparatus to which a voice code obtained by a first voice encoding method is input for converting this voice code to a voice code of a second voice encoding

10   method and outputting the latter voice code.

There has been an explosive increase in subscribers to cellular telephones in recent years and it is predicted that the number of such users will continue to grow in the future. Voice communication using the

15   Internet (Voice over IP, or VoIP) is coming into increasingly greater use in intracorporate IP networks (intranets) and for the provision of long-distance telephone service. In voice communication systems such as cellular telephone systems and VoIP, use is made of

20   voice encoding technology for compressing voice in order to utilize the communication line effectively. In the case of cellular telephones, the voice encoding technology used differs depending upon the country or system. With regard to W-CDMA expected to be employed

25   as the next-generation cellular telephone system, AMR (Adaptive Multi-Rate) has been adopted as the common global voice encoding method. With VoIP, on the other hand, a method compliant with ITU-T Recommendation

G.729A is being used as the voice encoding method. The AMR and G.729A methods both employ a basic algorithm referred to as CELP (Code Excited Linear Prediction). The CELP operating principles will now be described

5   taking the G.729A method as an example.

·CELP Operating Principles

CELP is characterized by the efficient transmission of linear prediction coefficients (LPC coefficients) representing the voice characteristics of the human

10 vocal tract, and a sound-source signal comprising the pitch component and noise component of voice. More specifically, in accordance with CELP, the human vocal tract is approximated by an LPC synthesis filter H(z) expressed by the following equation:

15
$$H(z) = \frac{1}{1 + \sum_{i=1}^{p} a_i z^{-i}} \qquad (1)$$

and it is assumed that the sound-source signal input to the LPC synthesis filter H(z) can be separated into a pitch-period component representing the periodicity of

20 voice and a noise component representing randomness. CELP, rather than transmitting the input voice signal to the decoder side directly, extracts the filter coefficients of the LPC synthesis filter and the pitch-period and noise components of the excitation signal,

25 quantizes these to obtain quantization indices and transmits the quantization indices, thereby implementing a high degree of information compression.

·Encoder structure and operation

Fig. 23 is a diagram illustrating a method
compliant with ITU-U Recommendation G.729A. As shown in
Fig. 23, input signals (voice signals) X of a
predetermined number (= N) of samples per frame are
5 input to an LPC analyzer 1 frame by frame. If the
sampling speed is 8 kHz and the period of a single frame
is 10 ms, then one frame is composed of 80 samples. The
LPC analyzer 1, which is regarded as an all-pole filter
represented by Equation (1), obtains filter coefficients
10 α_i (i = 1, $\cdots$, p), where p represents the order of the
filter. Generally, in the case of voice in the
telephone band, a value of 10 to 12 is used as p. The
LPC analyzer 1 performs LPC analysis using the input
signal (80 samples), 40 pre-read samples and 120 past
15 samples, for a total of 240 samples, and obtains the LPC
coefficients.

A parameter converter 2 converts the LPC
coefficients to LSP (Line Spectrum Pair) parameters. An
LSP parameter is a parameter of a frequency region in
20 which mutual conversion with LPC coefficients is
possible. Since a quantization characteristic is
superior to LPC coefficients, quantization is performed
in the LSP domain. An LSP quantizer 3 quantizes an LSP
parameter obtained by the conversion and obtains an LSP
25 code and an LSP dequantized value. An LSP interpolator
4 obtains an LSP interpolated value from the LSP
dequantized value found in the present frame and the LSP
dequantized value found in the previous frame. More

specifically, one frame is divided into two subframes, namely first and second subframes, of 5 ms each, and the LPC analyzer 1 determines the LPC coefficients of the second subframe but not of the first subframe. Using

5 the LSP dequantized value found in the present frame and the LSP dequantized value found in the previous frame, the LSP interpolator 4 predicts the LSP dequantized value of the first subframe by interpolation.

A parameter reverse converter 5 converts the LSP

10 dequantized value and the LSP interpolated value to LPC coefficients and sets these coefficients in an LPC synthesis filter 6. In this case, the LPC coefficients converted from the LSP interpolated values in the first subframe of the frame and the LPC coefficients converted

15 from the LSP dequantized values in the second subframe are used as the filter coefficients of the LPC synthesis filter 6. In subsequent description, 1 having subscript(s) is not a numeral, but an alphabet.

After LSP parameters LSPi (i = 1, $\cdots$, p) are

20 quantized as by scalar quantization or vector quantization in the LSP quantizer 3, the quantization indices (LSP codes) are sent to a decoder. Fig. 24 is a diagram useful in describing the quantization method. Here sets of large numbers of quantization LSP

25 parameters have been stored in a quantization table 3a in correspondence with index numbers 1 to n. A distance calculation unit 3b calculates distance in accordance with the following equation:

$$d = W \cdot \Sigma_i \{lsp_q(i) - lsp(i)\}^2 \qquad (i = 1 \sim p)$$

where W represents a weighting coefficient.

When q is varied from 1 to n, a minimum-distance index detector 3c finds the q for which the distance d is minimum and sends the index q to the decoder side as an LSP code.

Next, sound-source and gain search processing is executed. Sound source and gain are processed on a per-subframe basis. In accordance with CELP, a sound-source signal is divided into a pitch-period component and a noise component, an adaptive codebook 7 storing a sequence of past sound-source signals is used to quantize the pitch-period component and an algebraic codebook 8 or noise codebook is used to quantize the noise component. Described below will be typical CELP-type voice encoding using the adaptive codebook 7 and algebraic codebook 8 as sound-source codebooks.

The adaptive codebook 7 is adapted to output N samples of sound-source signals (referred to as "periodicity signals"), which are delayed successively by one sample, in association with indices 1 to L. Fig. 25 is a diagram showing the structure of the adaptive codebook 7 in the case of a subframe of 40 samples (N = 40). The adaptive codebook is constituted by a buffer BF for storing the pitch-period component of the latest (L+39) samples. A periodicity signal comprising 1 to 40 samples is specified by index 1, a periodicity signal comprising 2 to 41 samples is specified by index 2, $\cdots$,

and a periodicity signal comprising L to L+39 samples is specified by index L. In the initial state, the content of the adaptive codebook 7 is such that all signals have amplitudes of zero. Operation is such that a subframe

5 length of the oldest signals is discarded subframe by subframe so that the sound-source signal obtained in the present frame will be stored in the adaptive codebook 7.

An adaptive-codebook search identifies the periodicity component of the sound-source signal using

10 the adaptive codebook 7 storing past sound-source signals. That is, a subframe length (= 40 samples) of past sound-source signals in the adaptive codebook 7 are extracted while changing, one sample at a time, the point at which read-out from the adaptive codebook 7

15 starts, and the past sound-source signals are input to the LPC synthesis filter 6 to create a pitch synthesis signal βAP_L, where P_L represents a past periodicity signal (adaptive code vector), which corresponds to delay L, extracted from the adaptive codebook 7, A the

20 impulse response of the LPC synthesis filter 6, and β the gain of the adaptive codebook.

An arithmetic unit 9 finds an error power E_L between the input voice X and βAP_L in accordance with the following equation:

25 $$E_L = |X-\beta AP_L|^2 \qquad (2)$$

If we let AP_L represent a weighted synthesized signal output from the adaptive codebook, Rpp the autocorrelation of AP_L and Rxp the cross-correlation

between AP_L and the input signal X, then an adaptive code
vector P_L at a pitch lag Lopt for which the error power
of Equation (2) is minimum will be expressed by the
following equation:

5
$$P_L = argmax\left(\frac{R_{xp}^2}{R_{pp}}\right)$$
$$= argmax\left(\frac{(X^{\top}AP_L)^2}{(AP_L)^{\top}(AP_L)}\right) \qquad (3)$$

That is, the optimum starting point for read-out from
the adaptive codebook is that at which the value

10 obtained by normalizing the cross-correlation Rxp
between the weighted systhesized signal AP_L and the input
signal X by the autocorrelation Rpp of the weighted
systhesized signal is largest. Accordingly, an error-
power evaluation unit 10 finds the pitch lag Lopt that

15 satisfies Equation (3). Optimum pitch gain βopt is
given by the following equation:

βopt = Rxp/Rpp   (4)

Next, the noise component contained in the sound-
source signal is quantized using the algebraic codebook

20 8. The latter is constituted by a plurality of pulses
of amplitude 1 or -1. By way of example, Fig. 26
illustrates pulse positions for a case where frame
length is 40 samples. The algebraic codebook 8 divides
the N (= 40) sampling points constituting one frame into

25 a plurality of pulse-system groups 1 to 4 and, for all
combinations obtained by extracting one sampling point
from each of the pulse-system groups, successively
outputs, as noise components, pulsed signals having a +1

or a -1 pulse at each sampling point. In this example, basically four pulses are deployed per frame. Fig. 27 is a diagram useful in describing sampling points assigned to each of the pulse-system groups 1 to 4.

5   (1) Eight sampling points 0, 5, 10, 15, 20, 25, 30, 35 are assigned to the pulse-system group 1;

       (2) eight sampling points 1, 6, 11, 16, 21, 26, 31, 36 are assigned to the pulse-system group 2;

       (3) eight sampling points 2, 7, 12, 17, 22, 27, 32, 37 are assigned to the pulse-system group 3; and

       (4) 16 sampling points 3, 4, 8, 9, 13, 14, 18, 19, 23, 24, 28, 29, 33, 34, 38, 39 are assigned to the pulse-system group 4.

Three bits are required to express the sampling points in pulse-system groups 1 to 3 and one bit is required to express the sign of a pulse, for a total of four bits. Further, four bits are required to express the sampling points in pulse-system group 4 and one bit is required to express the sign of a pulse, for a total of five bits. Accordingly, 17 bits are necessary to specify a pulsed signal output from the algebraic codebook 8 having the pulse placement of Fig. 26, and 2^{17} ($= 2^4 \times 2^4 \times 2^4 \times 2^5$) types of pulsed signals exist.

The pulse positions of each of the pulse systems are limited as illustrated in Fig. 26. In the algebraic codebook search, a combination of pulses for which the error power relative to the input voice is minimized at the reproduction is decided from among the combinations

of pulse positions of each of the pulse systems. More specifically, with βopt as the optimum pitch gain found by the adaptive-codebook search, the output P_L of the adoptive codebook is multiplied by βopt and the product

5 is input to an adder 11. At the same time, the pulsed signals are input successively to the adder 11 from the algebraic codebook 8 and a pulse signal is specified that will minimize the difference between the input signal X and a reproduced signal obtained by inputting

10 the adder output to the LPC synthesis filter 6. More specifically, first a target vector X' for an algebraic codebook search is generated in accordance with the following equation using the optimum adaptive codebook output P_L and optimum pitch gain βopt obtained from the

15 input signal X by the adaptive-codebook search:

$$X' = X - \beta opt AP_L \qquad (5)$$

In this example, pulse position and amplitude (sign) are expressed by 17 bits and therefore 2^{17} combinations exist, as mentioned above. Accordingly,

20 letting C_K represent a kth algebraic-code output vector, a code vector C_K that will minimize an evaluation-function error output power D in the following equation is found by a search of the algebraic codebook:

$$D = |X'-G_cAC_K|^2 \qquad (6)$$

25 where G_c represents the gain of the algebraic codebook. Minimizing Equation (6) is equivalent to finding the C_K, i.e., the k, that will minimize the following equation:

$$D' = \frac{(X'^\mathsf{T} AC_k)^2}{(AC_k)^\mathsf{T}(AC_k)} \qquad (7)$$

Thus, in the algebraic codebook search, the error-power

evaluation unit 10 searches for the k that specifies the

combination of pulse position and polarity that will

5   afford the largest value obtained by normalizing the

cross-correlation between the algebraic synthesis signal

AC_K and target signal X' by the autocorrelation of the

algebraic synthesis signal AC_K.

    Gain quantization will be described next. With the

10   G.729A system, the algebraic codebook gain is not

quantized directly. Rather, the adaptive codebook gain

G_a (= βopt) and a correction coefficient γ of the

algebraic codebook gain G_c are vector quantized together.

The algebraic codebook gain G_c and the correction

15   coefficient γ are related as follows:

    G_c = g' × γ

where g' represents the gain of the present frame

predicted from the logarithmic gains of four past

subframes. A gain quantizer 12 has a gain quantization

20   table (gain codebook), not shown, for which there are

prepared 128 (= 2^7) combinations of adaptive codebook

gain G_a and correction coefficients γ for algebraic

codebook gain. The method of the gain codebook search

includes (1) extracting one set of table values from the

25   gain quantization table with regard to an output vector

from the adaptive codebook 7 and an output vector from

the algebraic codebook 8 and setting these values in

gain varying units 13, 14, respectively; (2) multiplying

these vectors by gains G_a, G_c using the gain varying units 13, 14, respectively, and inputting the products to the LPC synthesis filter 6; and (3) selecting, by way of the error-power evaluation unit 10, the combination

5   for which the error power relative to the input signal X is smallest.

A line encoder 15 creates line data by multiplexing (1) an LSP code, which is the quantization index of the LSP, (2) a pitch-lag code Lopt, (3) an algebraic code,

10   which is an algebraic codebook index, and (4) a gain code, which is a quantization index of gain, and sends the line data to the decoder.

Thus, as described above, the CELP system produces a model of the voice generation process, quantizes the

15   characteristic parameters of this model and transmits the parameters, thereby making it possible to compress voice efficiently.

·Decoder structure and operation

Fig. 28 is a block diagram illustrating a G.729A-

20   compliant decoder. Line data sent from the encoder side is input to a line decoder 21, which proceeds to output an LSP code, pitch-lag code, algebraic code and gain code. The decoder decodes voice data based upon these codes. The operation of the decoder will now be

25   described, though parts of the description will be redundant because functions of the decoder are included in the encoder.

Upon receiving the LSP code as an input, an LSP

dequantizer 22 applies dequantization and outputs an LSP

dequantized value. An LSP interpolator 23 interpolates

an LSP dequantized value of the first subframe of the

present frame from the LSP dequantized value in the

5   second subframe of the present frame and the LSP

dequantized value in the second subframe of the previous

frame. Next, a parameter reverse converter 24 converts

the LSP interpolated value and the LSP dequantized value

to LPC synthesis filter coefficients. A G.729A-

10   compliant synthesis filter 25 uses the LPC coefficient

converted from the LSP interpolated value in the initial

first subframe and uses the LPC coefficient converted

from the LSP dequantized value in the ensuing second

subframe.

15   An adaptive codebook 26 outputs a pitch signal of

subframe length (= 40 samples) from a read-out starting

point specified by a pitch-lag code, and a noise

codebook 27 outputs a pulse position and pulse polarity

from a read-out position that corresponds to an

20   algebraic code. A gain dequantizer 28 calculates an

adaptive codebook gain dequantized value and an

algebraic codebook gain dequantized value from the gain

code applied thereto and sets these vales in gain

varying units 29, 30, respectively. A adder 31 creates

25   a sound-source signal by adding a signal, which is

obtained by multiplying the output of the adaptive

codebook by the adaptive codebook gain dequantized

value, and a signal obtained by multiplying the output

of the algebraic codebook by the algebraic codebook gain
dequantized value. The sound-source signal is input to
an LPC synthesis filter 25. As a result, reproduced
voice can be obtained from the LPC synthesis filter 25.

5   In the initial state, the content of the adaptive
codebook 26 on the decoder side is such that all signals
have amplitudes of zero. Operation is such that a
subframe length of the oldest signals is discarded
subframe by subframe so that the sound-source signal

10   obtained in the present frame will be stored in the
adaptive codebook 26. In other words, the adaptive
codebook 7 of the encoder and the adaptive codebook 26
of the decoder are always maintained in the identical,
latest state.

15   ·Difference between G.729A and AMR-compliant
        encoding methods

        The difference between the G.729-compliant voice
encoding method and the AMR voice encoding method will
be described next. Fig. 29 illustrates results obtained

20   by comparing the main features of the G.729A and AMR
voice encoding methods. It should be noted that
although there are a total of eight types of AMR
encoding modes, the particulars shown in Fig. 29 are
common for all encoding modes. The G.729A and AMR voice

25   encoding methods have the same input-signal sampling
frequency (= 8 kHz), the same subframe length (= 5 ms)
and the same order of linear prediction (= order ten).
However, as shown in Fig. 30, they have different frame

lengths and different numbers of subframes per frame.
In the G.729A method, one frame is composed of two
subframes, namely 0th and 1st subframes; in the AMR
method, one frame is composed of four subframes, namely

5   0th to 3rd subframes.

Fig. 31 illustrates the result of comparing the bit
assignments of the G.729A and AMR methods. Fig. 31
illustrates a case where the mode for the AMR method is
7.95 kbps, which is nearest to the bit rate of the

10   G.729A method. It is obvious from Fig. 31 that although
the numbers of bits (= 17) of the algebraic codebook per
subframe are the same, the allocations of numbers of
bits necessary for other codes differ entirely.
Further, with the G.729A method, adaptive codebook gain

15   and algebraic codebook gain are vector quantized
collectively and, as a consequence, there is one type of
gain code per subframe. With the AMR method, however,
there are two types of gain codes, namely adaptive
codebook gain and algebraic codebook gain, per subframe.

20   As described above, a common basic algorithm is
used by the G.729A method now employed widely for VoIP
in the communication of voice over the Internet and by
the AMR method adopted for the next-generation cellular
telephone system. However, the frame lengths differ and

25   so do the numbers of bits expressing the codes.

It is believed that the growing popularity of the
Internet and cellular telephones will lead to ever
increasing voice traffic by Internet users and users of

cellular telephone networks. Fig. 32 is a conceptual view illustrating the relationship between networks and users in such case. In a case where a user A of a network (e.g., the Internet) 51 communicates by voice

5 with a user B of a network (e.g., a cellular telephone network) 53, communication between the users cannot take place if a first encoding method used in voice communication by the network 51 and a second encoding method used in voice communication by the network 53

10 differ.

Accordingly, a voice code converter 55 is provided between the networks, as shown in Fig. 32, and is adapted to convert the voice code that has been encoded by one network to the voice code of the encoding method

15 used in the other network.

Fig. 33 shows an example of the prior art using voice code conversion. This example takes into consideration only a case where voice input to a terminal 52 by user A is sent to a terminal 54 of user

20 B. It is assumed here that the terminal 52 possessed by user A has only an encoder 52a of an encoding method 1 and that the terminal 54 of user B has only a decoder 54a of an encoding method 2.

Voice that has been produced by user A on the

25 transmitting side is input to the encoder 52a of encoding method 1 incorporated in terminal 52. The encoder 52a encodes the input voice signal to a voice code of the encoding method 1 and outputs this code to a

transmission path 51'. When the voice code of encoding
method 1 enters via the transmission path 51', a decoder
55a of the voice code converter 55 decodes reproduced
voice from the voice code of encoding method 1. An

5 encoder 55b of the voice code converter 55 then converts
the reproduced voice signal to voice code of the
encoding method 2 and sends this voice code to a
transmission path 53'. The voice code of the encoding
method 2 is input to the terminal 54 through the

10 transmission path 53'. Upon receiving the voice code of
the encoding method 2 as an input, the decoder 54a
decodes reproduced voice from the voice code of the
encoding method 2. As a result, the user B on the
receiving side is capable of hearing the reproduced

15 voice. Processing for decoding voice that has first
been encoded and then re-encoding the decoded voice is
referred to as "tandem connection".

Voice (reproduced voice) consisting of information
compressed by encoding processing contains a lesser

20 amount of voice information in comparison with the
original voice (source) and, hence, the sound quality of
reproduced voice is inferior to that of the source. In
particular, with recent low-bit-rate voice encoding
typified by the G.729A and AMR methods, much information

25 contained in input voice is discarded in the encoding
process in order to realize a high compression rate.
When a tandem connection in which encoding and decoding
are repeated is employed, a problem which arises is a

marked decline in the quality of reproduced voice.

An additional problem with tandem processing is delay. It is known that when a delay in excess of 100 ms occurs in two-way communication such as a telephone

5 conversation, the delay is perceived by the communicating parties and is a hindrance to conversation. It is known also that even if real-time processing can be executed in voice encoding in which frame processing is carried out, a delay which is four

10 times the frame length basically is unavoidable. For example, since frame length in the AMR method is 20 ms, the delay is at least 80 ms. With the conventional method of voice code conversion, tandem connection is required in the G.729A and AMR methods. The delay in

15 such case is 160 ms or greater. Such a delay is perceivable by the parties in a telephone conversation and is an impediment to conversation.

As described above, in order for voice communication to be performed between networks employing

20 different voice encoding methods, the conventional practice is to execute tandem processing in which a compressed voice code is decoded into voice and then the voice code is re-encoded. Problems arise as a consequence, namely a pronounced decline in the quality

25 of reproduced voice and an impediment to telephone conversion caused by delay.

Another problem is that the prior art does not take the effects of transmission-path error into

consideration. More specifically, if wireless
communication is performed using a cellular telephone
and, bit error or burst error occurs owing to the
influence of phenomena such as phasing, the voice code

5 changes to one different from the original and there are
instances where the voice code of an entire frame is
lost. If traffic is heavy over the Internet,
transmission delay grows, the voice code of an entire
frame may be lost or frames may change places in terms

10 of their order. Since code conversion will be performed
based upon a voice code that is incorrect if
transmission-path error is a factor, a conversion to the
optimum voice code can no longer be achieved. Thus
there is need for a technique that will reduce the

15 effects of transmission-path error.

SUMMARY OF THE INVENTION

Accordingly, an object of the present invention is
to so arrange it that the quality of reconstructed voice
will not be degraded even when a voice code is converted

20 from that of a first voice encoding method to that of a
second voice encoding method.

Another object of the present invention is to so
arrange it that a voice delay can be reduced to improve
the quality of a telephone conversation even when a

25 voice code is converted from that of a first voice
encoding method to that of a second voice encoding
method.

Another object of the present invention is to

reduce a decline in the sound quality of reconstructed

voice ascribable to transmission-path error by

eliminating, to the maximum degree possible, the effects

of error from a voice code that has been distorted by

5   transmission-path error and applying a voice-code

conversion to the voice code in which the effects of

error have been reduced.

According to the present invention, the foregoing

objects are attained by providing a voice code

10   conversation apparatus to which a voice code obtained by

encoding performed by a first voice encoding method is

input for converting this voice code to a voice code of

a second voice encoding method, comprising: (1) code

separating means for separating codes of a plurality of

15   components necessary to reconstruct a voice signal from

the voice code based upon the first voice encoding

method; (2) dequantizers for dequantizing the codes of

each of the components and outputting dequantized

values; (3) quantizers for quantizing the dequantized

20   values, which are output from respective ones of the

dequantizers, by the second voice encoding method to

thereby generate codes; and (4) means for multiplexing

the codes output from respective ones of the quantizers

and outputting a voice code based upon the second voice

25   encoding method.

In accordance with the voice code conversion

apparatus according to the present invention, a voice

code based upon a first voice encoding method is

dequantized and the dequantized values are quantized and encoded by a second voice encoding method. As a consequence, there is no need to output reconstructed voice in the voice code conversion process. This means that it is possible to suppress a decline in the quality of voice that is eventually reproduced and to reduce signal delay by shortening processing time.

According to another aspect of the present invention, there is provided an acoustic code conversion apparatus to which an acoustic code obtained by encoding an acoustic signal by a first encoding method frame by frame is input for converting this acoustic code to an acoustic code of a second encoding method and outputting the latter acoustic code, comprising: (1) code separating means for separating codes of a plurality of components necessary to reconstruct an acoustic signal from the acoustic code based upon the first encoding method; (2) dequantizers for dequantizing the codes of each of the components and outputting dequantized values if a transmission-path error has not occurred, and outputting dequantized values obtained by applying error concealment processing if a transmission-path error has occurred; (3) quantizers for quantizing the dequantized values, which are output from respective ones of the dequantizers, by the second encoding method to thereby generate codes; and (4) means for multiplexing the codes output from respective ones of the quantizers and outputting an acoustic code that is based upon the

second encoding method.

Other features and advantages of the present invention will be apparent from the following description taken in conjunction with the accompanying

5 drawings, in which like reference characters designate the same or similar parts throughout the figures thereof.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram illustrating the

10 principles of the present invention;

Fig. 2 is a block diagram illustrating the principles of the present invention in greater detail;

Fig. 3 is a block diagram of a first embodiment of the present invention;

15 Figs. 4A and 4B are diagrams useful in describing frames subjected to LSP quantization;

Fig. 5 is a block diagram illustrating the construction of an LSP quantizer;

Fig. 6 is a diagram showing the correspondence

20 between frames and subframes;

Figs. 7A and 7B are diagrams showing the relationship between pitch lag and indices in the G.729A method;

Figs. 8A and 8B are diagrams showing the

25 relationship between pitch lag and indices in the AMR method;

Fig. 9 is a diagram showing the corresponding relationship between pitch lag in the G.729A method and

pitch lag in the AMR method;

Fig. 10 is a block diagram illustrating the construction of a gain quantizer;

Fig. 11 is a diagram of a second embodiment of the present invention;

Fig. 12 is a block diagram illustrating the construction of an LSP quantizer according to the second embodiment;

Fig. 13 is a flowchart of SLP encoding processing according to the second embodiment;

Fig. 14 is part one of a processing flowchart according to a third embodiment;

Fig. 15 is part two of a processing flowchart according to the third embodiment;

Fig. 16 is a diagram of a fourth embodiment of the present invention;

Fig. 17 is a diagram useful in describing processing executed by an LSP code converter according to the fourth embodiment;

Fig. 18 is a block diagram of an LSP dequantizer;

Fig. 19 is a block diagram of an LSP quantizer;

Fig. 20 is a block diagram useful in describing the intrusion of channel error into voice code;

Fig. 21 is a block diagram useful in describing the principles of a fifth embodiment of the present invention;

Fig. 22 is a block diagram showing the fifth embodiment.

Fig. 23 is a block diagram of an encoder based upon ITU-T G.729A code conversion according to the prior art;

Fig. 24 is a diagram useful in describing a quantization method according to the prior art;

5   Fig. 25 is a diagram useful in describing an adaptive codebook according to the prior art;

Fig. 26 is a diagram useful in describing an algebraic codebook used in G.729A code conversion according to the prior art;

10   Fig. 27 is a diagram useful in describing sampling points of pulse-system groups according to the prior art;

Fig. 28 is a block diagram of a decoder based upon ITU-T G.729A code conversion according to the prior art;

15   Fig. 29 is a diagram showing a comparison of main features of ITU-T G.729A code conversion and AMR code conversion according to the prior art;

Fig. 30 is a diagram showing a comparison of frame lengths according to the prior art;

20   Fig. 31 is a diagram showing a comparison of bit allocations in ITU-T G.729A code conversion and AMR code conversion according to the prior art;

Fig. 32 is a conceptual view of the prior art; and

Fig. 33 illustrates an example of voice code

25   conversion according to the prior art.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

(A) Principles of the present invention

Fig. 1 is a block diagram illustrating the

principles of a voice code conversion apparatus according to the present invention. The apparatus receives, as an input signal, a voice code obtained by a first voice encoding method (encoding method 1), and

5   converts this voice code to a voice code of a second voice encoding method (encoding method 2).

An encoder 61a of encoding method 1 incorporated in a terminal 61 encodes a voice signal produced by user A to a voice code of encoding method 1 and sends this

10   voice code to a transmission path 71. A voice code conversion unit 80 converts the voice code of encoding method 1 that has entered from the transmission path 71 to a voice code of encoding method 2 and sends this voice code to a transmission path 72. A decoder 91a in

15   a terminal 91 decodes reproduced voice from the voice code of encoding method 2 that enters via the transmission path 72, and a user B is capable of hearing the reproduced voice.

The encoding method 1 encodes a voice signal by (1)

20   a first LPC code obtained by quantizing linear prediction coefficients (LPC coefficients), which are obtained by frame-by-frame linear prediction analysis, or LSP parameters found from these LPC coefficients; (2) a first pitch-lag code, which specifies the output

25   signal of an adaptive codebook that is for outputting a periodic sound-source signal; (3) a first noise code, which specifies the output signal of a noise codebook that is for outputting a noisy sound-source signal; and

(4) a first gain code obtained by collectively
quantizing adaptive codebook gain, which represents the
amplitude of the output signal of the adaptive codebook,
and noise codebook gain, which represents the amplitude

5 of the output signal of the noise codebook. The
encoding method 2 encodes a voice signal by (1) a second
LPC code, (2) a second pitch-lag code, (3) a second
noise code and (4) a second gain code, which are
obtained by quantization in accordance with a

10 quantization method different from that of the encoding
method 1.

The voice code conversion unit 80 has a code
separator 81, an LSP code converter 82, a pitch-lag code
converter 83, an algebraic code converter 84, a gain

15 code converter 85 and a code multiplexer 86. The code
separator 81 separates the voice code of the encoding
method 1, which code enters from the encoder 61a of
terminal 61 via the transmission path 71, into codes of
a plurality of components necessary to reproduce a voice

20 signal, namely (1) LSP code, (2) pitch-lag code, (3)
algebraic code and (4) gain code. These codes are input
to the code converters 82, 83, 84 and 85, respectively.
The latter convert the entered LSP code, pitch-lag code,
algebraic code and gain code of the encoding method 1 to

25 LSP code, pitch-lag code, algebraic code and gain code
of the encoding method 2, and the code multiplexer 86
multiplexes these codes of the encoding method 2 and
sends the multiplexed signal to the transmission path

72.

Fig. 2 is a block diagram illustrating the voice
code conversion unit in which the construction of the
code converters 82 t0 85 is clarified. Components in
Fig. 2 identical with those shown in Fig. 1 are
designated by like reference characters. The code
separator 81 separates an LSP code 1, a pitch-lag code
1, an algebraic code 1 and a gain code 1 from line data
(the voice signal based upon encoding method 1) that
enters from the transmission path via an input terminal
#1, and inputs these codes to the code converters 82,
83, 84 and 85, respectively.

The LSP code converter 82 has an LSP dequantizer
82a for dequantizing the LSP code 1 of encoding method 1
and outputting an LSP dequantized value, and an LSP
quantizer 82b for quantizing the LSP dequantized value
by the encoding method 2 and outputting an LSP code 2.
The pitch-lag code converter 83 has a pitch-lag
dequantizer 83a for dequantizing the pitch-lag code 1 of
encoding method 1 and outputting a pitch-lag dequantized
value, and a pitch-lag quantizer 83b for quantizing the
pitch-lag dequantized value by the encoding method 2 and
outputting a pitch-lag code 2. The algebraic code
converter 84 has an algebraic dequantizer 84a for
dequantizing the algebraic code 1 of encoding method 1
and outputting an algebraic dequantized value, and an
algebraic quantizer 84b for quantizing the algebraic
dequantized value by the encoding method 2 and

outputting an algebraic code 2. The gain code converter 85 has a gain dequantizer 85a for dequantizing the gain code 1 of encoding method 1 and outputting a gain dequantized value, and a gain quantizer 85b for

5 quantizing the gain dequantized value by the encoding method 2 and outputting a gain code 2.

The code multiplexer 86 multiplexes the LSP code 2, pitch-lag code 2, algebraic code 2 and gain code 2, which are output from the quantizers 82b, 83b, 84b and

10 85b, respectively, thereby creating a voice code based upon the encoding method 2 and sends this voice code to the transmission path from an output terminal #2.

In the prior art, the input is a reproduced voice obtained by decoding a voice code that has been encoded

15 in accordance with encoding method 1, and, the reproduced voice is encoded again in accordance with encoding method 2 and then is decoded. As a consequence, since voice parameters are extracted from reproduced voice in which the amount of information has

20 been reduced slightly in comparison with the source owing to the re-encoding (i.e., voice-information compression), the voice code obtained thereby is not necessarily the optimum voice code. By contrast, in accordance with the code conversion apparatus of the

25 present invention, the voice code of encoding method 1 is converted to the voice code of encoding method 2 via the process of dequantization and quantization. As a result, it is possible to carry out voice code

conversion with much less degradation in comparison with the conventional tandem connection. An additional advantage is that since it is unnecessary to effect decoding into voice in order to perform the voice code conversion, there is little of the delay that is a problem with the conventional tandem connection.

(B) First embodiment

Fig. 3 is a block diagram illustrating a voice code conversion unit according to a first embodiment of the present invention. Components identical with those shown in Fig. 2 are designated by like reference characters. This arrangement differs from that Fig. 2 in that a buffer 87 is provided and in that the gain quantizer of the gain code converter 85 is constituted by an adaptive codebook gain quantizer $85b_1$ and a noise codebook gain quantizer $85b_2$. Further, in the first embodiment shown in Fig. 3, it is assumed that the G.729A encoding method is used as encoding method 1 and the AMR method as the encoding method 2. Though there are eight encoding modes in AMR encoding, in this embodiment use is made of an encoding mode having a transmission rate of 7.95 kbps.

As shown in Fig. 3, an nth frame of channel data bst1(n) is input to terminal #1 from a G.729A encoder (not shown) via the transmission path. Here the bit rate of G.729A encoding is 8 kbps and therefore the line data bst1(n) is represented by bit sequence of 80 bits. The code separator 81 separates LSP code I_LSP1(n),

pitch-lag code I_LAG1(n,j), algebraic code I_CODE1(n,j) and gain code I_GAIN1(n,j) from the line data bst1(n) and inputs these codes to the converters 82, 83, 84 and 85, respectively. The suffix j represents the number of

5   the 0th and 1st subframes constituting each frame and takes on values of 0 and 1.

(a)   LSP code converter

Figs. 4A and 4B are diagrams illustrating the relationship between frames and LSP quantization in the

10   G.729A and AMR encoding methods. As shown in Fig. 4A, frame length according to the G.729A method is 10 ms and an LSP parameter found from the input voice signal of the first subframe (1st subframe) is quantized only once every 10 ms. By contrast, frame length according to the

15   AMR method is 20 ms and an LSP parameter is quantized from the input signal of the third subframe (3rd subframe)only once every 20 ms. In other words, if the same 20 ms is considered to be the frame length in both cases, the G.729A method performs LSP quantization twice

20   whereas the AMR method performs quantization only once. This means that the LSP codes of two consecutive frames in the G.729A method cannot be converted as it is to the LSP code of the AMR method.

According to the first embodiment, therefore, it is

25   so arranged that only the LSP codes of odd-numbered frames are converted to LSP codes of the AMR method; the LSP codes of even-numbered frames are not converted. It is also possible, however, to adopt an arrangement in

which only the LSP codes of even-numbered frames are converted to LSP codes of the AMR method and the LSP codes of the odd-numbered frames are not converted. Further, as will be described below, the G.729A-compliant LSP dequantizer 82a uses interframe prediction and therefore the updating of status is performed frame by frame.

When LSP code I_LSP1(n+1) of an odd-numbered frame enters the LSP dequantizer 82a, the latter dequantizes the code and outputs LSP dequantized values lsp(i) (i = 1, $\cdots$, 10). Here the LSP dequantizer 82a performs the same operation as that of a dequantizer used in a decoder of the G.729A encoding method.

Next, when an LSP dequantized value lsp(i) enters the LSP quantizer 82b, the latter quantizes the value in accordance with the AMR encoding method and obtains LSP code I_LSP2(m). Here it is not necessarily required that the LSP quantizer 82b be exactly the same as the quantizer used in an encoder of the AMR encoding method, although it is assumed that at least the LSP quantization table thereof is the same as that of the AMR encoding method.

·LSP dequantization method in LSP dequantizers

The G.729A-compliant LSP dequantization method used in the LSP dequantizer 82a will be described in line with G.729. If LSP code I_SLP1(n) of an nth frame is input to the LSP dequantizer 82a, the latter divides this code into four codes L_0, L_1, L_2 and L_3. The code L_1

represents an element number (index number) of a first

LSP codebook CB1, and the codes L_2, L_3 represent element

numbers of second and third LSP codebooks CB_2, CB_3,

respectively. The first LSP codebook CB1 has 128 sets

5   of 10-dimensional vectors, and the second and third LSP

codebooks CB2 and CB3 both have 32 sets of 5-dimensional

vectors. The code L_0 indicates which of two types of MA

prediction coefficients (described later) to use.

Next, a residual vector $l_i^{(n)}$ of the nth frame is

10   found by the following equation:

$$l_i^{(n)} = \begin{cases} CB1(L_1, i) + CB2(L_2, i) & (i = 1, \ldots, 5) \\ CB1(L_1, i) + CB3(L_3, i - 5) & (i = 6, \ldots, 10) \end{cases} \qquad (8)$$

A residual vector $l_i^{(n+1)}$ of the (n+1)th frame can be

found in similar fashion. An LSF coefficient $\omega(i)$ is

15   found from the residual vector $l_i^{(n+1)}$ of the (n+1)th

frame and residual vectors $l_i^{(n+1-k)}$ of the past four

frames in accordance with the following equation:

$$\omega(i) = (1 - \sum_{k=1}^{4} p(i, k))l_i^{(n+1)} + \sum_{k=1}^{4} p(i, k)l_i^{(n+1-k)} \qquad , (i = 1, \ldots, 10) \qquad (9)$$

20   where p(i,k) represents which coefficient of the two

types of MA prediction coefficients has been specified

by the code L_0. It should be noted that an LSF

coefficient is not found from a residual vector with

regard to the nth frame. The reason for this is that

25   the nth frame is not quantized by the LSP quantizer.

However, the residual vector $l_i^{(n)}$ is necessary to update

status.

Next, the LSP dequantizer 82a finds an LSP

dequantized value lsp(i) from the LSP coefficient ω(i) using the following equation:

$$lsp(i) = \cos[\omega(i)] \quad (i = 1, \cdots, 10) \quad (10)$$

·LSP quantization method in LSP quantizers

5   The details of the LSP quantization method used in the LSP quantizer 82b will now be described. In accordance with the AMR encoding method, a common LSP quantization method is used in seven of the eight modes, with the 12.2-kbps mode being excluded. Only the size 10 of the LSP codebook differs. The LSP quantization method in the 7.95-kbps mode will be described here.

      If the LSP dequantized value lsp(i) has been found by Equation (10), the LSP quantizer 82b finds a residual vector $r(i)^{(m)}$ by subtracting a prediction vector $q(i)^{(m)}$ 15 from the LSP dequantized value lsp(i) in accordance with the following equation:

$$r(i)^{(m)} = lsp(i) - q(i)^{(m)} \qquad \cdots (11)$$

where m represents the number of the present frame.

      The prediction vector $q(i)^{(m)}$ is found in accordance 20 with the following equation using a quantized residual vector $\hat{r}(i)^{(m-1)}$ of the immediately preceding frame and an MA prediction vector a(i):

$$q(i)^{(m)} = a(i)\hat{r}(i)^{(m-1)} \qquad (12)$$

In accordance with the AMR encoding method, the 10- 25 dimensional vector $r(i)^{(m)}$ is divided into three small vectors $r_1(i)$ (i = 1, 2, 3), $r_2(i)$ (i = 4, 5, 6) and $r_3(i)$ (i = 7, 8, 9, 10) and each of these small vectors is vector-quantized using nine bits.

Vector quantization is so-called pattern-matching processing. From among prepared codebooks (codebooks for which the dimensional lengths are the same as those of the small vectors) CB1 to CB3, the LSP quantizer 82b

5 selects a codebook for which the weighted Euclidean distance to each small vector is minimum. The selected codebook serves as the optimum codebook vector. Let I_1, I_2 and I_3 represent numbers (indices) indicating the particular element numbers of the optimum codebook

10 vector in each of the codebooks CB1 to CB3. The LSP quantizer 82b outputs an LSP code I_LSP2(m) obtained by combining these indices I_1, I_2, I_3. Since the sizes of all of the codebooks CB1 to CB3 are nine bits (512 sets), the word length of each of the indices I_1, I_2, I_3

15 also is nine bits and the LSP code I_LSP2(m) has a word length that is a total of 27 bits.

Fig. 5 is a block diagram illustrating the construction of the LSP quantizer 82b. The latter includes a residual vector calculation unit $82b_1$ for

20 outputting the following 10-dimensional residual vector in accordance with Equation (11):

$$r(i) = r_1(i) \ (i=1,2,3), \ r_2(i) \ (i=4,5,6), \ r_3(i)$$

$$(i=7,8,9,10)$$

Optimum codebook vector decision units $82b_2$ to $82b_4$

25 output the index numbers I_1, I_2, I_3 of an optimum codebook vector for which the weighted Euclidean distances to each of the small vectors $r_1(i)$ $(i=1,2,3)$, $r_2(i)$ $(i=4,5,6)$, $r_3(i)$ $(i=7,8,9,10)$ are minimum.

Further, 512 sets of 3-dimensional low-frequency LSP vectors r(j,1), r(j,2), r(j,3) (i = 1 ~ 512) have been stored in the low-frequency LSP codebook CB1 of the optimum codebook vector decision unit $82b_2$ in correspondence with indices 1 ~ 512. A distance calculation unit DSC calculates distance in accordance with the following equation:

$$d = \Sigma_i \{r(j,i) - r_1(i)\}^2 \quad (i = 1 \sim 3)$$

When j is varied from 1 to 512, a minimum distance index detector MDI finds the j for which distance d is minimized and outputs j as an LSP code I_1 for the low order region.

Though the details are not shown, the optimum codebook vector decision units $82b_3$ and $82b_4$ use the midrange-frequency LSP codebook CB2 and high-frequency LSP codebook CB3 to output the indices I_2 and I_3, respectively, in a manner similar to that of the optimum codebook vector decision unit $82b_2$.

(b) Pitch-lag code converter

The pitch-lag code converter 83 will now be described.

As mentioned above (see Fig. 29), frame length is 10 ms in the G.729A encoding method and 20 ms in the AMR encoding method. When the pitch-lag code is converted, therefore, it is necessary that pitch-lag code of two frames in the G.729A method be converted as one frame of pitch-lag code in the AMR method.

Consider a case where pitch-lag codes of the nth

and (n+1)th frames in the G.729A method are converted to pitch-lag code of the mth frame in the AMR method. If it is assumed that the leading timing of the nth frame in the G.729A method and the leading timing of the mth frame in the AMR method are equal, then the relationship between the frames and subframes of the G.729A and AMR methods will be as shown in (a) of Fig. 6. Further, the quantization bit numbers of pitch lag in each subframe of the G.729A and AMR methods will be as shown in (b) of Fig. 6 (see Fig. 31).

In even-numbered subframes, therefore, the methods of encoding pitch-lags in the G.729A and AMR are exactly the same and the numbers of quantization bits are the same, i.e., eight. This means that a G.729A-compliant pitch-lag code can be converted to an AMR-compliant pitch-lag code in regard to even-numbered subframes by the following equations:

$$I\_LAG2(m,0) = I\_LAG1(n,0) \qquad (13)$$

$$I\_LAG2(m,2) = I\_LAG1(n+1,0) \qquad (14)$$

On the other hand, in odd numbered subframes, quantization of the difference between integral lag of the present frame and integral lag of the preceding subframe is performed in the G.729A and AMR. Since the number of quantization bits is one larger for the AMR method, the conversion can be made by the following equations:

$$I\_LAG2(m,1) = I\_LAG1(n,1) + 15 \qquad (15)$$

$$I\_LAG2(m,3) = I\_LAG1(n+1,1) + 15 \qquad (16)$$

Equations (13), (14) and Equations (15), (16) will be described in greater detail.

With the G.729A and AMR methods, pitch lag is decided assuming that the pitch period of voice is
5   between 2.5 and 18 ms. If the pitch lag is an integer, encoding processing is simple. In the case of a short pitch period, however, frequency resolution is unsatisfactory and voice quality declines. For this reason, a sample interpolation filter is used to decide
10   pitch lag at one-third the sampling precision in the G.729A and AMR methods. That is, it is just as if a voice signal sampled at a period that is one-third the actual sampling period has been stored in the adaptive codebook.
15   Thus, two types of pitch lag exist, namely integral lag indicating the actual sampling period and non-integral lag indicating one-third the sampling period.

Figs. 7A and 7B illustrate the relationship between pitch lag and indices in the G.729A method, in which
20   Fig. 7A shows the case for even-numbered subframes and Fig. 7B the case for odd-numbered subframes. In the case of the odd-numbered frames, indices are assigned at one-third the sampling precision for lag values in the range 19+1/3 to 85 and at a precision of one sample for
25   lag values in the range 85 to 143. Here the integral portion of lag is referred to as "integral lag" and the non-integral portion (fractional portion) is referred to as "non-integral lag". In the G.729A method, eight bits

are assigned to the pitch lag of the even-numbered

subframes and, hence, there are 256 pitch-lag indices.

For example, index is 4 in a case where pitch-lag is

20+2/3 and index is 254 in a case where is 142.

5   On the other hand, in the case of odd-numbered

subframes according to the G.729A method, the difference

between integral lag T_{old} of the previous subframe (even-

numbered) and pitch lag (integral pitch lag or non-

integral pitch lag) of the present subframe is quantized

10   using five bits (32 patterns). In the case of odd-

numbered subframes, it is assumed that T_{old} is a

reference point and that the index of T_{old} is 17, as

shown in Fig. 7B. It is assumed that the index of lag

that is 5+2/3 samples smaller than T_{old} is zero and that

15   the index of lag that is 4+2/3 samples larger than T_{old}

is 31. In other words, the range $T_{old} - (5+2/3)$ to $T_{old} +$

(4+2/3) is equally divided at one-third sample intervals

and indices of 32 patterns (5 bits) are assigned.

The relationship between pitch lag and indices in

20   the AMR method will now be described. Fig. 8 is a

diagram illustrating the relationship between pitch lag

and indices in the AMR method, in which Fig. 8A shows

the case for even-numbered subframes and Fig. 8B the

case for odd-numbered subframes. In the case of the

25   even-numbered subframes of the AMR method, eight bits

are assigned to the pitch-lag indices. Pitch lag is

composed of integral lag and non-integral lag, and the

index-number assignment method is exactly the same as

that of the G.729A method. Accordingly, in the case of even-numbered subframes, G.729A-compliant pitch-lag indices can be converted to AMR-compliant pitch-lag indices by Equations (13) and (14).

5   On the other hand, in the case of odd-numbered subframes according to the AMR method, the difference between integral lag T_{old} of the previous subframe and pitch lag of the present subframe is quantized just as in the case of the G.729A method. However, the number

10   of quantization bits is one larger than in the case of the G.729A method and quantization is performed using six bits (64 patterns). In the case of odd-numbered subframes, it is assumed that T_{old} is a reference point and that the index of T_{old} is 32, as shown in Fig. 8B.

15   It is assumed that the index of lag that is 10+2/3 samples smaller than T_{old} is zero and that the index of lag that is 9+2/3 samples larger than T_{old} is 63. In other words, the range T_{old} - (10+2/3) to T_{old} + (9+2/3) is equally divided at one-third sample intervals and

20   indices of 64 patterns (6 bits) are assigned.

Fig. 9 is a diagram of corresponding relationships in a case where G.729A-compliant indices in odd-numbered subframes are converted to AMR-compliant indices. As will be understood from Fig. 9, the indices are shifted

25   by a total of 15 from the G.729A method to the AMR method even though the lag values are the same. For example, with regard to lag -(5+2/3), the 0th index is assigned in the G.729A method but the 15th index is

assigned in the AMR method. Accordingly, in order to convert G.729A-compliant indices in odd-numbered subframes to AMR-compliant indices, it is necessary to correct the index values by adding on 15, as indicated

5 by Equations (15), (16).

(c) Conversion of algebraic code

Conversion of algebraic code will be described next.

Although frame length in the G.729A method differs

10 from that in the AMR method, subframe length is the same for both, namely 5 ms (40 samples). In other words, the relationship between frames and subframes in the G.729A and AMR methods is as illustrated in (a) of Fig. 6. Further, the numbers of quantization bits of algebraic

15 code in each subframe of the G.729A and AMR methods is as illustrated in (c) of Fig. 6 (see Fig. 31). Furthermore, the algebraic codebooks of both methods have the structure depicted in Fig. 26; the structures are exactly the same.

20 Accordingly, the four pulse positions and the pulse polarity information that are the results output from the algebraic codebook search in the G.729A method can be replaced as it is on a one-to-one basis by the results output from the algebraic codebook search in the

25 AMR method. The algebraic-code conversions are as indicated by the following:

$$I\_CODE2(m,0) = I\_CODE1(n,0) \qquad (17)$$

$$I\_CODE2(m,1) = I\_CODE1(n,1) \qquad (18)$$

$$I\_CODE2(m,2) = I\_CODE1(n+1,0) \qquad (19)$$

$$I\_CODE2(m,3) = I\_CODE1(n+1,1) \qquad (20)$$

(d)　Conversion of gain code

Conversion of gain code will be described next.

First, gain code $I\_GAIN(n,0)$ is input to the gain dequantizer 85a (Fig. 3). In accordance with the G.729A method, vector quantization is used to quantize the gain. Accordingly, an adaptive codebook gain dequantized value G_a and a dequantized value γ_c of a correction coefficient for algebraic codebook gain are sought as the gain dequantized value. The algebraic codebook gain is found in accordance with the following equation using a prediction value g_c', which is predicted from the logarithmic energy of algebraic codebook gain of the past four subframes, and γ_c:

$$G_c = g_c'\gamma_c \qquad (21)$$

In the AMR method, adaptive codebook gain G_a and algebraic codebook gain G_c are quantized separately and therefore quantization is performed separately by the adaptive codebook gain quantizer $85b_1$ and algebraic codebook gain quantizer $85b_2$ of the AMR method in the gain code converter 85. It is unnecessary to identify the adaptive codebook gain quantizer $85b_1$ and the algebraic codebook gain quantizer $85b_2$ with those used by AMR method. But, at least an adaptive codebook gain table and an algebraic codebook table of the quantizers $85b_1$, $85b_2$ are same as those used by AMR method.

Fig. 10 is a diagram showing the constructions of

the adaptive codebook gain quantizer 85b$_1$ and algebraic codebook gain quantizer 85b$_2$.

First, the adaptive codebook gain dequantized value G$_a$ is input to the adaptive codebook gain quantizer 85b$_1$ and is subjected to scalar quantization. Values G$_a$(i) (i = 1 ~ 16) of 16 types (four bits) which are the same as those of the AMR method have been stored in a scalar quantization table SQTa. A squared-error calculation unit ERCa calculates the square of the error between the adaptive codebook gain dequantized value G$_a$ and each table value, i.e., $[G_a-G_a(i)]^2$, and an index detector IXDa obtains, as the optimum value, the table value that minimizes the error that prevails when i is varied from 1 to 16, and outputs this index as adaptive codebook gain code I_GAIN2a(m,0) in the AMR method.

Next, Gc, which is found in accordance with Equation (21) from the noise codebook gain dequantized value γ_c and g$_c$', is input to the algebraic codebook gain quantizer 85b$_2$ in order to undergo scalar quantization. Values G$_c$(i) (i = 1 ~ 32) of 32 types (five bits) which are the same as those of the AMR method have been stored in a scalar quantization table SQTc. A squared-error calculation unit ERCc calculates the square of the error between the noise codebook gain dequantized value G$_c$ and each table value, i.e., $[G_c-G_c(i)]^2$, and an index detector IXDc obtains, as the optimum value, the table value that minimizes the error that prevails when i is varied from 1 to 32, and outputs this index as noise

codebook gain code I_GAIN2c(m,0) in the AMR method.

Similar processing is thenceforth executed to find AMR-compliant adaptive codebook gain code I_GAIN2a(m,1) and noise codebook gain code I_GAIN2c(m,1) from G.729A-compliant gain code I_GAIN1(n,1).

5

Similarly, AMR-compliant adaptive codebook gain code I_GAIN2a(m,2) and noise codebook gain code I_GAIN2c(m,2) are found from G.729A-compliant gain code I_GAIN1(n+1,0), and AMR-compliant adaptive codebook gain code I_GAIN2a(m,3) and noise codebook gain code

10

I_GAIN2c(m,3) are found from G.729A-compliant gain code I_GAIN1(n+1,1).

(e) Code transmission processing

The buffer 87 of Fig. 3 holds the codes output from

15

the converters 82 to 85 until the processing of two frames of G.729A code (one frame in the AMR method) is completed. The converted code is then input to the code multiplexer 86. When all of the codes of one frame in the AMR method have been acquired, the code multiplexer

20

86 multiplexes the code data, converts the data to line data and sends the line data to the transmission path from the output terminal #2.

Thus, in accordance with the first embodiment, as described above, G.729A-compliant voice code can be

25

converted to AMR-compliant voice code without being decoded into voice. As a result, delay can be reduced over that encountered with the conventional tandem connection and a decline in sound quality can be reduced

as well.

(C)   Second embodiment

Fig. 11 is a diagram useful in describing an

overview of a second embodiment of the present

5   invention. The second embodiment improves upon the LSP

quantizer 82b in the LSP code converter 82 of the first

embodiment; the overall arrangement of the voice code

conversion unit is the same as that of the first

embodiment (Fig. 3).

10   Fig. 11 illustrates a case where LSP code of nth

and (n+1)th frames of the G.729A method is converted to

LSP code of the mth frame of the AMR method. In Fig.

11, LSP0(i) (i = 1, $\cdots$, 10) represent 10-dimensional

LSP dequantized values in a first subframe (1st subframe)

15   of an nth frame according to the G.729A method, and

LSP1(i) (i = 1, $\cdots$, 10) represent 10-dimensional LSP

dequantized values in a first subframe (1st subframe) of

an (n+1)th frame according to the G.729A method.

Further, old_LSP(i) (i = 1, $\cdots$, 10) represent 10-

20   dimensional LSP dequantized values in a 1st subframe of a

past frame [(n-1)th frame].

In a case where voice code is converted from the

G.729A to the AMR method, a dequantized value LSP0(i) in

the G.729A method is not converted to an LSP code in the

25   AMR method because of the difference in frame length, as

pointed out in the first embodiment. In other words, an

LSP is quantized one time per frame in the G.729A method

and therefore LSP0(i), LSP1(i) are quantized together

and sent to the decoder side. In order to convert voice
code from the G.729A to the AMR method, however, it is
necessary to encode and convert LSP parameters in
conformity with the operation of the AMR-compliant

5   decoder. As a consequence, the dequantized value
LSP1(i) in the G.729A method is converted to
AMR-compliant code but the dequantized value LSP0(i) is
not converted to AMR-compliant code.

According to the AMR method, one frame consists of

10   four subframes and only the LSP parameters of the final
subframe (3rd subframe) are quantized and transmitted.
In the decoder, therefore, LSP parameters LSPc0(i),
LSPc1(i) and LSPc2(i) of the 0th, 1st and 2nd subframes
are found from the dequantized value old_LSPc(i) of the

15   previous frame and the LSP parameter LSPc3(i) of the 3rd
subframe in the present frame in accordance with the
following interpolation equations:

$$LSPc0(i) = 0.75 \ old\_LSPc(i) + 0.25 \ LSPc3(i)$$
$$(i = 1, 2, \ldots 10) \qquad (22)$$

20   $$LSPc1(i) = 0.50 \ old\_LSPc(i) + 0.50 \ LSPc3(i)$$
$$(i = 1, 2, \ldots 10) \qquad (23)$$

$$LSPc2(i) = 0.25 \ old\_LSPc(i) + 0.75 \ LSPc3(i)$$
$$(i = 1, 2, \ldots 10) \qquad (24)$$

If the quality of input voice does not change

25   abruptly, which is the case with voiced sounds, the LSP
parameters also do not change abruptly. This means that
no particular problems arise even if an LSP dequantized
value is converted to code so as to minimize the LSP

quantization error in the final subframe (3rd subframe),
as in the first embodiment, and the LSP parameters of
the other 0th to 3rd subframes are found by the
interpolation operations of Equations (22) to (24).

5   However, if voice quality changes suddenly, as in the
case of unvoiced or transient segments, and, more
particularly, if the quality of voice changes suddenly
within the frame, there are instances where the
conversion method of the first embodiment is

10   unsatisfactory. Accordingly, in the second embodiment,
code conversion is carried out taking into consideration
not only LSP quantization error in the final subframe
but also interpolation error stemming from LSP
interpolation.

15   When a dequantized value LSP1(i) is converted to
AMR-compliant LSP code according to the first
embodiment, the conversion is made using as a reference
only the square of the error between the LSP parameter
LSPc3(i), which is specified by the above-mentioned LSP

20   code, and the dequantized value LSP1(i). By contrast,
in the second embodiment, encoding is performed taking
into consideration not only the above-mentioned square
of the error but also the square of the error between
the dequantized value LSP0(i) and the LSP parameter

25   LSPc1(i) obtained by the interpolation operation of
Equation (23).

        Fig. 12 is a diagram illustrating the construction
of the LSP quantizer 82b according to the second

embodiment, and Fig. 13 is a flowchart of conversion

processing according to the second embodiment. Each LSP

vector (LSP parameter) of the ten dimensions will be

considered upon dividing it into three small vectors of

5   a low-frequency region (first to third dimensions), a

midrange-frequency region (fourth to sixth dimensions)

and a high-frequency region (seventh to tenth

dimensions).

·Processing for deciding low-frequency three-

10   dimensional LSP codes

First, the processing set forth below is executed

with regard to the small vector of the low-frequency

region (three dimensions of the low-frequency region)

among the values $LSP1(i)$ ($i = 1, \ldots 10$). The LSP

15   codebooks used here are of three types, namely the low-

frequency codebook CB1 (3 dimensions × 512 sets), the

midrange-frequency codebook CB2 (3 dimensions × 512

sets) and the high-frequency codebook CB3 (4 dimensions

× 512 sets).

20   A residual vector calculation unit DBC calculates a

residual vector $r_1(i)$ ($i = 1 \sim 3$) by subtracting a

prediction vector from the low-frequency LSP dequantized

value $LSP1(i)$ ($i = 1 \sim 3$) (step 101).

Next, a processing unit (CPU) performs the

25   operation $I_1 = 1$ (step 102), extracts an I_1th code vector

$CB1(I_1,i)$ ($i = 1 \sim 3$) from the low-frequency codebook CB1

(step 103), finds a conversion error $E_1(I_1)$ between this

code vector and the residual vector $r_1(i)$ ($i = 1 \sim 3$) in

accordance with the following equation:

$$E_1(I_1) = \sum_i \{r_1(i) - CB1(I_1,i)\}^2 \quad (i = 1 \sim 3)$$

and stores this error in a memory MEM (step 104).

Next, using Equation (23), the CPU interpolates

5 LSPc1(i) (i = 1 ~ 3) from the LSP dequantized value

LSPc3(i) (i = 1 ~ 3), which prevailed when the code

vector CB1(I_1,i) was selected, and the preceding

dequantized value old_LSPc(i) (i = 1 ~ 3) (step 105),

calculates the conversion error $E_2(I_1)$ between LSP0(i)

10 and LSPc1(i) in accordance with the following equation:

$$E_2(I_1) = \sum_i \{LSP0(i) - LSPc1(i)\}^2 \quad (i = 1 \sim 3)$$

and stores this error in the memory MEM (step 106).

Next, using the following equation, the CPU

calculates an error $E(I_1)$ that prevailed when the I_1th

15 code vector was selected and stores this error in memory

(step 107):

$$E(I_1) = E_1(I_1) + E_2(I_1)$$

The CPU then compares the error $E(I_1)$ with a minimum

error $minE(I_1)$ thus far (step 108) and updates the error

20 $E(I_1)$ to $minE(I_1)$ if $E(I_1) < minE(I_1)$ holds (step 109).

Following update processing, the CPU checks to see

whether I_1 = 512 holds (step 110). If I_1 < 512 holds,

the CPU increments I_1 (I_1 + 1 → I_1; step 111). The CPU

then repeats processing from step 103 onward. If I_1 =

25 512 holds, however, the CPU decides, as the low-

frequency three-dimensional LSP code, the index I_1 for

which the error $E(I_1)$ is minimized (step 112).

·Processing for deciding midrange-frequency

three-dimensional LSP codes

If processing for deciding the low-frequency three-dimensional LSP code I_1 is completed, the CPU executes the processing set forth below with regard to the small

5   vector (three-dimensional) of the midrange-frequency region.

The residual vector calculation unit DBC calculates a residual vector $r_2(i)$ ($i = 4 \sim 6$) by subtracting a prediction vector from the midrange-frequency LSP

10   dequantized value LSP1(i) ($i = 4 \sim 6$).

Next, the processing unit (CPU) performs the operation $I_2 = 1$, extracts an I_2th code vector $CB2(I_2,i)$ ($i = 4 \sim 6$) from the midrange-frequency codebook CB2, finds a conversion error $E_1(I_2)$ between this code vector

15   and the residual vector $r_2(i)$ ($i = 4 \sim 6$) in accordance with the following equation:

$$E_1(I_2) = \sum_i \{r_2(i) - CB2(I_2,i)\}^2 \quad (i = 4 \sim 6)$$

and stores this error in the memory MEM.

Next, using Equation (23), the CPU interpolates

20   LSPc1(i) ($i = 4 \sim 6$) from the LSP dequantized value LSPc3(i) ($i = 4 \sim 6$), which prevailed when the code vector $CB2(I_2,i)$ was selected, and the preceding dequantized value old_LSPc(i) ($i = 4 \sim 6$), calculates the conversion error $E_2(I_2)$ between LSP0(i) and LSPc1(i)

25   in accordance with the following equation:

$$E_2(I_2) = \sum_i \{LSP0(i) - LSPc1(i)\}^2 \quad (i = 4 \sim 6)$$

and stores this error in the memory MEM.

Next, using the following equation, the CPU

calculates an error $E(I_2)$ that prevailed when the I_2th code vector was selected and stores this error in memory:

$$E(I_2) = E_1(I_2) + E_2(I_2)$$

5   The CPU then compares the error $E(I_2)$ with a minimum error $minE(I_2)$ thus far and updates the error $E(I_2)$ to $minE(I_2)$ if $E(I_2) < minE(I_2)$ holds.

Following update processing, the CPU checks to see whether $I_2 = 512$ holds. If $I_2 < 512$ holds, the CPU

10   increments I_2 ($I_2 + 1 \rightarrow I_2$). The CPU then repeats the above-described processing. If $I_2 = 512$ holds, however, the CPU decides, as the midrange-frequency three-dimensional LSP code, the index I_2 for which the error $E(I_2)$ is minimized.

15   ·Processing for deciding high-frequency four-
               dimensional LSP codes

If processing for deciding the midrange-frequency three-dimensional LSP code I_2 is completed, the CPU executes the processing set forth below with regard to

20   the small vector (four-dimensional) of the high-frequency region.

The residual vector calculation unit DBC calculates a residual vector $r_3(i)$ ($i = 7 \sim 10$) by subtracting a prediction vector from the high-frequency LSP

25   dequantized value $LSP1(i)$ ($i = 7 \sim 10$).

Next, the processing unit (CPU) performs the operation $I_3 = 1$, extracts an I_3th code vector $CB3(I_3, i)$ ($i = 7 \sim 10$) from the high-frequency codebook CB3, finds

a conversion error $E_1(I_3)$ between this code vector and the residual vector $r_3(i)$ ($i = 7 \sim 10$) in accordance with the following equation:

$$E_1(I_3) = \Sigma_i \{r_3(i) - CB3(I_3,i)\}^2 \quad (i = 7 \sim 10)$$

5   and stores this error in the memory MEM.

Next, using Equation (23), the CPU interpolates $LSPc1(i)$ ($i = 7 \sim 10$) from the LSP dequantized value $LSPc3(i)$ ($i = 7 \sim 10$), which prevailed when the code vector $CB3(I_3,i)$ was selected, and the preceding

10   dequantized value $old\_LSPc(i)$ ($i = 7 \sim 10$), calculates the conversion error $E_2(I_3)$ between $LSP0(i)$ and $LSPc1(i)$ in accordance with the following equation:

$$E_2(I_3) = \Sigma_i \{LSP0(i) - LSPc1(i)\}^2 \quad (i = 7 \sim 10)$$

and stores this error in the memory MEM.

15   Next, using the following equation, the CPU calculates an error $E(I_3)$ that prevailed when the I_3th code vector was selected and stores this error in memory:

$$E(I_3) = E_1(I_3) + E_2(I_3)$$

20   The CPU then compares the error $E(I_3)$ with a minimum error $minE(I_3)$ thus far and updates the error $E(I_3)$ to $minE(I_3)$ if $E(I_3) < minE(I_3)$ holds.

Following update processing, the CPU checks to see whether $I_3 = 512$ holds. If $I_3 < 512$ holds, the CPU

25   increments I_3 ($I_2 + 1 \rightarrow I_2$). The CPU then repeats the above-described processing. If $I_3 = 512$ holds, however, the CPU decides, as the high-frequency four-dimensional LSP code, the index I_3 for which the error $E(I_3)$ is

minimized.

Thus, in the second embodiment, the conversion error of LSPc1(i) is taken into account as interpolator error. However, it is also possible to decide the LSP code upon taking the conversion error of LSPc0(i) and LSPc2(i) into account in similar fashion.

Further, in the second embodiment, the description assumes that the weightings of E_1 and E_2 are equal as the error evaluation reference. However, the LSP code can also be decided upon so arranging it that E_1 and E_2 are weighted separately as $E = \omega_1 E_1 + \omega_2 E_2$.

Thus, in accordance with the second embodiment, as described above, a G.729A-compliant voice code can be converted to AMR-compliant code without being decoded to voice. As a result, delay can be reduced over that encountered with the conventional tandem connection and a decline in sound quality can be reduced as well. Moreover, not only conversion error that prevails when LSP1(i) is re-quantized but also interpolation error due to the LSP interpolator are taken into consideration. This makes it possible to perform an excellent voice code conversion with little conversion error even in a case where the quality of input voice varies within the frame.

(D)   Third embodiment

The third embodiment improves upon the LSP quantizer 82b in the LSP code converter 82 of the second embodiment. The overall arrangement is the same as that

of the first embodiment shown in Fig. 3.

The third embodiment is characterized by making a preliminary selection (selection of a plurality of candidates) for each of the small vectors of the low-,

5 midrange- and high-frequency regions, and finally deciding a combination $\{I_1, I_2, I_3\}$ of LSP code vectors for which the errors in all bands will be minimal. The reason for this approach is that there are instances where the 10-diminsional LSP synthesized code vector

10 synthesized from code vectors for which the error is minimal in each band is not the optimum vector. In particular, since an LPC synthesis filter is composed of LPC coefficients obtained by conversion from 10-dimensional LSP parameters in the AMR or G.729A method,

15 the conversion error in the LSP parameter region exerts great influence upon reproduced voice. Accordingly, it is desirable not only to perform a codebook search for which error is minimized for each small vector of the LSP but also to finally decide LSP code that will

20 minimize error (distortion) of 10-dimensional LSP parameters obtained by combining small vectors.

Figs. 14 and 15 are flowcharts of conversion processing executed by the LSP quantizer 82b according to the third embodiment. Here the LSP quantizer 82b has

25 the same block components as those shown in Fig. 12; only the processing executed by the CPU is different.

The 10-dimensional dequantized value output from the LSP dequantizer 82a is divided into three areas,

namely a low-frequency 3-dimensional small vector

LSP1(i) (i = 1 ~ 3), a midrange-frequency 3-dimensional

small vector LSP1(i) (i = 4 ~ 6) and a high-frequency

four-dimensional small vector LSP1(i) (i = 7 ~ 10) (step

5   201).

Next, the residual vector calculation unit DBC

calculates a residual vector $r_1(i)$ (i = 1 ~ 3) by

subtracting a prediction vector from the low-frequency

LSP dequantized value LSP1(i) (i = 1 ~ 3) (step 202).

10   The processing unit (CPU) then performs the operation I_1

= 1 (step 203), extracts an I_1th code vector CB1(I_1,i) (i

= 1 ~ 3) from the low-frequency codebook CB1 (step 204),

finds a conversion error $E_1(I_1)$ between this code vector

and the residual vector $r_1(i)$ (i = 1 ~ 3) in accordance

15   with the following equation:

$$E_1(I_1) = \sum_i \{r_1(i) - CB1(I_1,i)\}^2 \quad (i = 1 ~ 3)$$

and stores this error in the memory MEM (step 205).

Next, using Equation (23), the CPU interpolates

LSPc1(i) (i = 1 ~ 3) from the LSP dequantized value

20   LSPc3(i) (i = 1 ~ 3), which prevailed when the code

vector CB1(I_1,i) was selected, and the preceding

dequantized value old_LSPc(i) (i = 1 ~ 3) (step 206),

calculates the conversion error $E_2(I_1)$ between LSP0(i)

and LSPc1(i) in accordance with the following equation:

25   $$E_2(I_1) = \sum_i \{LSP0(i) - LSPc1(i)\}^2 \quad (i = 1 ~ 3)$$

and stores this error in the memory MEM (step 207).

Next, using the following equation, the CPU

calculates an error $E_L(I_1)$ that prevailed when the I_1th

code vector was selected and stores this error in memory
(step 208):

$$E_L(I_1) = E_1(I_1) + E_2(I_1)$$

The processor thenceforth checks to see whether $I_1 = 512$
holds (step 209). If $I_1 < 512$ holds, the CPU increments
I_1 ($I_1 + 1 \rightarrow I_1$; step 210). The CPU then repeats
processing from step 204 onward. If $I_1 = 512$ holds,
however, the CPU selects N_L-number of code-vector
candidates starting from the smaller ones of $E_L(I_1)$ ($I_1 =$
1 ~ 512) and adopts $PSEL_{I1}(j)$ ($j = 1, \ldots N_L$) as the
index of each of the candidates (step 211).

If processing for deciding the low-frequency three-
dimensional small vector is completed, the CPU executes
similar processing with regard to the midrange-frequency
three-dimensional small vector. Specifically, the CPU
calculates 512 sets of errors $E_M(I_2)$ (step 212) by
processing similar to that of steps 202 to 210. Next,
the CPU selects N_M-number of code-vector candidates from
the smaller ones of $E_M(I_2)$ ($I_2 = 1$ ~ 512) and adopts
$PSEL_{I2}(k)$ ($k = 1, \ldots N_M$) as the index of each candidate
(step 213).

If processing for deciding the midrange-frequency
three-dimensional small vector is completed, the CPU
executes similar processing with regard to the high-
frequency four-dimensional small vector. Specifically,
the CPU calculates 512 sets of errors $E_H(I_3)$ (step 214),
selects N_H-number of code-vector candidates from the
smaller ones of $E_H(I_3)$ ($I_3 = 1$ ~ 512) and adopts $PSEL_{I3}(m)$

$(m = 1, \ldots N_H)$ as the index of each candidate (step 215).

A combination for which the errors in all bands will be minimal is decided by the following processing from the candidates that were selected by the processing set forth above: Specifically, the CPU finds the combined error

$$E(j,k,m) = E_L[PSEL_{I1}(j)] + E_M[PSEL_{I2}(k)] +$$
$$E_H[PSEL_{I3}(m)]$$

that prevails when $PSEL_{I1}(j)$, $PSEL_{I2}(k)$, $PSEL_{I3}(m)$ are selected from the N_L-number of low-frequency, N_M-number of midrange-frequency and N_H-number of high-frequency index candidates that were selected by the above-described processing (step 216), decides the combination, from among all combinations of j, k, m, for which the combined error $E(j,k,m)$ will be minimum, and outputs the following indices, which prevail at this time, as the LSP codes of the AMR method (step 217):

$$PSEL_{I1}(j), PSEL_{I2}(k), PSEL_{I3}(m)$$

According to the third embodiment, the conversion error of $LSPc1(i)$ is taken into account as interpolator error. However, it is also possible to decide the LSP code upon taking the conversion error of $LSPc0(i)$ and $LSPc2(i)$ into account in similar fashion.

Further, in the third embodiment, the description assumes that the weightings of E_1 and E_2 are equal as the error evaluation reference. However, the LSP code can also be decided upon so arranging it that E_1 and E_2 are

weighted separately as $E = \omega_1 E_1 + \omega_2 E_2$.

Thus, in accordance with the third embodiment, as described above, a G.729A-compliant voice code can be converted to AMR-compliant code without being decoded to voice. As a result, delay can be reduced over that encountered with the conventional tandem connection and a decline in sound quality can be reduced as well. Moreover, not only conversion error that prevails when LSP1(i) is re-quantized but also interpolation error due to the LSP interpolator are taken into consideration. This makes it possible to perform an excellent voice code conversion with little conversion error even in a case where the quality of input voice varies within the frame.

Further, the third embodiment is adapted to find a combination of code vectors for which combined error in all bands will be minimal from combinations of code vectors selected from a plurality of code vectors of each band, and to decide LSP code based upon the combination found. As a result, this embodiment can provide reproduced voice having a sound quality superior to that of the second embodiment.

(E)   Fourth embodiment

The foregoing embodiment relates to a case where the G.729A encoding method is used as the encoding method 1 and the AMR encoding method is used as the encoding method 2. In a fourth embodiment, the 7.95-kbps mode of the AMR encoding method is used as the

encoding method 1 and the G.729A encoding method is used as the encoding method 2.

Fig. 16 is a block diagram illustrating a voice code conversion unit according to a fourth embodiment of the present invention. Components identical with those shown in Fig. 2 are designated by like reference characters. This arrangement differs from that Fig. 2 in that the buffer 87 is provided and in that the gain dequantizer of the gain code converter 85 is constituted by an adaptive codebook gain dequantizer $85a_1$ and a noise codebook gain dequantizer $85a_2$. Further, in Fig. 16, the 7.95-kbps mode of the AMR method is used as the encoding method 1 and the G.729A encoding method is used as the encoding method 2.

As shown in Fig. 16, an mth frame of line data bst1(m) is input to terminal #1 from an AMR-compliant encoder (not shown) via the transmission path. Here the bit rate of AMR encoding is 7.95 kbps and the frame length is 20ms and therefore the line data bst1(m) is represented by bit sequence of 159 bits. The code separator 81 separates LSP code I_LSP1(m), pitch-lag code I_LAG1(m,j), algebraic code I_CODE1(m,j), adaptive codebook gain code I_GAIN1a(m,j) and algebraic codebook gain code I_GAIN1c(m,j) from the line data bst1(n) and inputs these codes to the converters 82, 83, 84, 85. The suffix j represents the four subframes constituting each frame in the AMR method and takes on any of the values 0, 1, 2 and 3.

(a) LSP code converter

·Overview of LSP code conversion processing

As shown in Fig. 4B, frame length according to the

AMR method is 20 ms and an LSP parameter is quantized

5   from the input signal of the third subframe only once

every 20 ms. By contrast, frame length according to the

G.729A method is 10 ms and an LSP parameter found from

the input voice signal of the first subframe is

quantized only once every 10 ms. Accordingly, two

10   frames of LSP code in the G.729A method must be created

from one frame of LSP code in the AMR method.

Fig. 17 is a diagram useful in describing

conversion processing executed by the LSP code converter

82 according to the fourth embodiment.

15   The LSP dequantizer 82a dequantizes LSP code

I_LSP1(m) of the third subframe in the mth frame of the

AMR method and generates a dequantized value $lsp_m(i)$.

Further, using the dequantized value $lsp_m(i)$ and a

dequantized value $lsp_{m-1}(i)$ of the third subframe in the

20   (m-1)th frame, which is the previous frame, the LSP

dequantizer 82a predicts a dequantized value $lsp_c(i)$ of

the first subframe in the mth frame by interpolation.

The LSP quantizer 82b quantizes the dequantized value

$lsp_c(i)$ of the first subframe in the mth frame in

25   accordance with the G.729A method and outputs LSP code

I_LSP2(n) of the first subframe of the nth frame.

Further, the LSP quantizer 82b quantizes the dequantized

value $lsp_m(i)$ of the third subframe in the mth frame in

accordance with the G.729A method and outputs LSP code I_LSP2(n+1) of the first subframe of the (n+1)th frame in the G.729A method.

·LSP dequantization

5   Fig. 18 is a diagram showing the construction of the LSP dequantizer 82a.

The LSP dequantizer 82a has 9-bit (512-pattern) codebooks CB1, CB2, CB3 for each of the small vectors when the AMR-method 10-dimensional LSP parameters are
10   divided into the small vectors of first to third dimensions, fourth to sixth dimensions and seventh to tenth dimensions. The LSP code I_LSP1(m) of the AMR method is decomposed into codes I_1, I_2, I_3 and the codes are input to the residual vector calculation unit DBC.
15   The code I_1 represents the element number (index) of the low-frequency 3-dimensional codebook CB1, and the codes I_2, I_3 also represent the element numbers (indices) of the midrange-frequency 3-dimensional codebook CB2 and high-frequency 4-dimensional codebook CB3, respectively.
20   Upon being provided with LSP code I_LSP1(m) = $\{I_1, I_2, I_3\}$, a residual vector creation unit DBG extracts code vectors corresponding to the codes I_1, I_2, I_3 from the codebooks CB1, CB2, CB3 and arrays the code vectors in the order of the codebooks CB1 ~ CB3 as follows:
25    $r(i,1)$ ~ $r(i,3)$, $r(i,4)$ ~ $r(i,6)$, $r(i,7)$ ~ $r(i,10)$, to create a 10-dimensional vector $r(i)^{(m)}$ ($i = 1, \ldots$ 10). Since prediction is used when LSP parameters are encoded in the AMR method, $r(i)^{(m)}$ is the vector of a

residual area. Accordingly, an LSP dequantized value lspm(i) of an mth frame can be found by adding a residual vector $r(i)^{(m)}$ of the present frame to a vector obtained by multiplying a residual vector $r(i)^{(m-1)}$ of the

5   previous frame by a constant p(i). That is, a dequantized-value calculation unit RQC calculates the LSP dequantized value lspm(i) in accordance with the following equation:

$$lsp_m(i) = r(i)^{(m-1)} \cdot p(i) + r(i)^{(m)} \qquad (25)$$

10   It should be noted that the constant p(i) used to multiply the residual vector $r(i)^{(m-1)}$ employs one that has been decided for every index i by the specifications of the AMR encoding method.

Next, using an LSP dequantized value $lsp_{m-1}(i)$ found

15   in the previous (m-1)th frame and $lsp_m(i)$ of the mth frame, a dequantized-value interpolator RQI obtains an LSP dequantized value $lsp_c(i)$ of the first frame in the mth frame by interpolation. Though any interpolation method may be used, the method indicated by the

20   following equation is used by way of example:

$$lsp_c(i) = \frac{lsp_{m-1}(i) + lsp_m(i)}{2}, \quad (i = 1, \dots, 10) \qquad (26)$$

By virtue of the foregoing, the LSP dequantizer 82a calculates and outputs dequantized values $lsp_m(i)$, $lsp_c(i)$ of the first and third subframes in the mth

25   frame.

·LSP quantization

LSP code I_LSP2(n) corresponding to the first subframe of the nth frame in the G.729A encoding method

can be found by quantizing the LSP parameter $lsp_c(i)$, which has been interpolated in accordance with Equation (26), through the method set forth below. Further, LSP code I_LSP2(n+1) corresponding to the first subframe of the (n+1)th frame in the G.729A encoding method can be found by quantizing $lsp_m(i)$ through a similar method.

First, the LSP dequantized value $lsp_c(i)$ is converted to an LSF coefficient $\omega(i)$ by the following equation:

$$\omega(i) = \arccos[lsp_c(i)], \quad (i = 1, \ldots, 10) \quad (27)$$

This is followed by quantizing, using 17 bits, residual vectors obtained by subtracting predicted components (predicted components obtained from codebook outputs of the past four frames) from the LSF coefficients $\omega(i)$.

In accordance with G.729A encoding, three codebooks cb1 (ten dimensional and seven bits), cb2 (five dimensions and five bits) and cb3 (five dimensions and five bits) are provided. Predicted components $\hat{l}^{(n-1)}$, $\hat{l}^{(n-2)}$, $\hat{l}^{(n-3)}$, $\hat{l}^{(n-4)}$ are found from each of the codebook outputs of the past four frames in accordance with the following equations:

$$\hat{l}_i^{(n-k)} = \begin{cases} cb1(L_1(n-k),i) + cb2(L_2(n-k),i) & (i=1,\ldots,5) \\ cb1(L_1(n-k),i) + cb3(L_3(n-k),i-5) & (i=6,\ldots,10) \end{cases} \quad (28)$$

where L1(n-k) represents the code (index) of codebook cb1 in the (n-k)th frame and cb1[L1(n-k)] is assumed to be a code vector (output vector) indicated by the index $L_1(n-k)$ of codebook cb1 in the (n-k)th frame. The same holds true for $L_2(n-k)$ and $L_3(n-k)$. Next, a residual

vector l_i (i = 1, ..., 10) is found by the following

equation:

$$l_i = \frac{\omega_i - \sum_{k=1}^{4} p(i,k) \, \tilde{l}_i^{(m-k)}}{1 - \sum_{k=1}^{4} p(i,k)} \qquad (29)$$

5   where p(i,k) is referred to as a prediction coefficient

and is a constant determined beforehand by the

specifications of the G.729A encoding method. The

residual vector l_i is what undergoes vector quantization.

Vector quantization is executed as follows: First,

10   codebook cb1 is searched to decide the index (code) L_1 of

the code vector for which the mean-square error is

minimum. Next, the 10-dimensional code vector

corresponding to the index L_1 is subtracted from the 10-

dimensional residual vector I_i to create a new target

15   vector. The codebook cb2 is searched in regard to the

lower five dimensions of the new target vector to decide

the index (code) L_2 of the code vector for which the

mean-square error is minimum. Similarly, the codebook

cb3 is searched in regard to the higher five dimensions

20   of the new target vector to decide the index (code) L_3 of

the code vector for which the mean-square error is

minimum. The 17-bit code that can be formed by arraying

these obtained codes L_1, L_2, L_3 as bit sequences is

output as LSP code L_LSP2(n) in the G.729A encoding

25   method. The LSP code I_LSP2(n+1) in the G.729A encoding

method can be obtained through exactly the same method

with regard to the LSP dequantized value lsp_m(i) as well.

Fig. 19 is a diagram showing the construction of

the LSP quantizer 82b. The residual vector calculation
unit DBC calculates the residual vectors in accordance
with Equations (27) to (29). A first codebook cb1 of a
first encoder CD1 has 128 sets (seven bits) of 10-

5   dimensional code vectors. A distance calculation unit
DSC1 calculates 128 sets of squared errors (Euclidean
distances) between residual vectors l_i (i = 1 ~ 10) and
code vectors $l(L_1,i)$ (i = 1 ~ 10), and an index detector
IXD1 detects and outputs the index L_1 of the code vector

10   for which the error is minimum from among the L_1 = 1 ~
128 code vectors. A subtractor SBC subtracts the 10-
dimensional vectors (L_1,i) (i = 1 ~ 10) corresponding to
the index L_1 of the first codebook cb1 from the 10-
dimensional residual vectors l_i (i = 1 ~ 10) and creates

15   a new target vector l_i' (i = 1 ~ 10). A second encoder
CD2 searches the codebook cb2 in regard to the lower
five dimensions l' (i = 1 ~ 5) of the new target vector
to decide the index (code) L_2 of the code vector l'
(L_2,i) (i = 1 ~ 5) for which the mean-square error is

20   minimum. Similarly, the third encoder CD3 searches the
codebook cb3 in regard to the higher five dimensions l_i'
(i = 6 ~ 10) of the new target vector to decide the
index (code) L_3 of the code vector l' (L_3,i) (i = 6 ~ 10)
for which the mean-square error is minimum.

25   (b)   Pitch-lag code converter
         Pitch-lag code conversion will be described next.
         With the G.729A and AMR encoding methods, pitch lag
is decided at one-third the sampling precision using a

sample interpolation filter, as set forth in connection
with the first embodiment. For this reason, two types
of lag, namely integral lag and non-integral lag, exist.
The relationship between pitch lag and indices in the

5   G.729A method is as illustrated in Figs. 7A and 7B and
need not be described again as it is identical with that
of the first embodiment. Further, the relationship
between pitch lag and indices in the AMR method is as
illustrated in Figs. 8A and 8B and need not be described

10 again as it is identical with that of the first
embodiment.

Accordingly, the methods of quantizing pitch lag
and the numbers of quantization bits are exactly the
same in the AMR and G.729A methods with regard to even-

15 numbered subframes. This means that pitch-lag indices
of even-numbered subframes in the AMR method can be
converted to pitch-lag indices of 0th subframes in two
consecutive frames of the G.729A method in accordance
with the following equations:

20   I_LAG2(n,0) = I_LAG1(m,0)   (30)

        I_LAG2(n+1,0) = I_LAG1(m,2)   (31)

With regard to odd-numbered subframes, a point in
common is that the difference between integral lag T_{old}
in the previous subframe and pith lag in the present

25 subframe is quantized. With respect to the number (six)
of quantization bits in the AMR method, the number is
smaller than that (five) in the G.729A method. This
makes necessary the following expedient:

First, integral lag Int(m,1) and non-integral lag Frac(m,1) are found from lag code I_LAG1(m,1) of the first subframe of the mth frame in the ATM method and the pitch lag is found by the following equation:

5 $$P = Int(m,1) + Frac(m,1)$$

The integral lag and non-integral lag corresponding to the indices (lag codes) are in one-to-one correspondence. If there are 28 lag codes, for example, then integral lag will be -1, non-integral lag will be

10 -1/3, and pitch lag P will be -(1+1/3), as illustrated in Fig. 8B.

Next, it is determined whether the pitch lag P found falls within the 5-bit pitch-lag range $T_{old}-(5+2/3)$ to $T_{old}-(4+2/3)$ in the G.729A odd-numbered subframes

15 shown in Fig. 7B. This range will be expressed as $[T_{old}-(5+2/3), T_{old}+(4+2/3)]$ below. If the corresponding relationship between AMR-compliant pitch lag and indices and the corresponding relationship between G.729A-compliant pitch lag and indices in the odd-numbered

20 subframes are compared, it will be found that there is a shift of 15 indices, as described earlier in connection with the first embodiment. Accordingly, if pitch lag P falls within the above-mentioned pitch-lag range, a correction is applied in accordance with the following

25 equations:

$$I\_LAG2(n,1) = I\_LAG1(m,1) - 15 \qquad (32)$$

$$I\_LAG2(n+1,1) = I\_LAG1(m,3) - 15 \qquad (33)$$

As a result, pitch lag I_LAG1(m,1) in the AMR method can

be converted to pitch lag I_LAG2(n,1) in the G.729A

method. Similarly, pitch lag I_LAG1(m,3) in the AMR

method can be converted to pitch lag I_LAG2(n+1,1) in

the G.729A method.

5   If pitch lag P does not fall within the above-

mentioned pitch-lag range, then pitch lag is clipped.

That is, if pitch lag P is smaller than $T_{old}-(5+2/3)$,

e.g., if pitch lag P is equal to $T_{old}-7$, then pitch lag P

is clipped to $T_{old}-(5+2/3)$. If pitch lag P is greater

10   than $T_{old}+(4+2/3)$, e.g., if pitch lag P is equal to

$T_{old}+7$, then pitch lag P is clipped to $T_{old}+(4+2/3)$.

        Though it may appear at a glance that such clipping

of pitch lag will invite a decline in voice quality,

preliminary experiments by the Inventors have

15   demonstrated that there is almost no decline in sound

quality even if such clipping processing is applied. It

is known that in such voiced segments as "ah" and "ee",

pitch lag varies smoothly and fluctuation in pitch lag P

in voiced odd-numbered subframes is small, falling

20   within the range $T_{old}-(5+2/3)$, $T_{old}+(4+2/3)$ in most cases.

In fluctuating segments such as rising or falling

segments, on the other hand, the value of pitch lag P

exceeds the above-mentioned range. However, in segments

where the quality of voice varies, the influence of the

25   adaptive codebook on reconstructed voice derived from a

periodic sound source declines. Hence there is almost

no influence on the quality of sound even when clipping

processing is executed. In accordance with the method

described above, AMR-compliant pitch-lag code can be converted to G.729A-compliant pitch-lag code.

(c) Conversion of algebraic code

The conversion of algebraic code will be described next.

Though frame length in the AMR method differs from that in the G.729A method, subframe length is the same 5 ms (40 samples) and the structure of the algebraic code is exactly the same in both methods. Accordingly, the four pulse positions and the pulse polarity information that are results output from the algebraic codebook search in the AMR method can be replaced as is on a one-to-one basis by the results output from the algebraic codebook search in the G.729A method. The algebraic-code conversions, therefore, are as indicated by the following:

$$I\_CODE2(n,0) = I\_CODE1(m,0) \qquad (34)$$

$$I\_CODE2(n,1) = I\_CODE1(m,1) \qquad (35)$$

$$I\_CODE2(n+1,0) = I\_CODE1(m,2) \qquad (36)$$

$$I\_CODE2(n+1,1) = I\_CODE1(m,3) \qquad (37)$$

(d) Conversion of gain code

Conversion of gain code will be described next.

First, adaptive codebook gain code $I\_GAINa(m,0)$ of the 0th subframe in the mth frame of the AMR method is input to the adaptive codebook gain dequantizer $85a_1$ to obtain the adaptive codebook gain dequantized value G_a. In accordance with the G.729A method, vector quantization is used to quantize the gain. The adaptive

codebook gain dequantizer $85a_1$ has a 4-bit (16-pattern) adaptive codebook gain table the same as that of the AMR method and refers to this table to output the adaptive codebook gain dequantized value G_a that corresponds to

5   the code I_GAIN1a(m,0).

Next, adaptive codebook gain code I_GAINc(m,0) of the 0th subframe in the mth frame of the AMR method is input to the noise codebook gain dequantizer $85a_2$ to obtain the algebraic codebook gain dequantized value G_c.

10   In the AMR method, interframe prediction is used in the quantization of algebraic codebook gain., gain is predicted from the logarithmic energy of algebraic codebook gain of the past four subframes and the correction coefficients thereof are quantized. To

15   accomplish this, the noise codebook gain dequantizer $85a_2$, which has a 5-bit (32-pattern) correction coefficient table the same as that of the AMR method, finds a table value γ_c of a correction coefficient that corresponds to the code I_GAIN1c(m,0) and outputs the

20   dequantized value $G_c = (g_c' \times \gamma_c)$ of algebraic codebook gain. It should be noted that the gain prediction method is exactly the same as the prediction method performed by the AMR-compliant decoder.

Next, the gains G_a, G_c are input to the gain

25   quantizer 85b to effect a conversion to G.729A-compliant gain code. The gain quantizer 85b uses a 7-bit gain quantization table the same as that of the G.729A method. This quantization table is two-dimensional, the

first element thereof is adaptive codebook gain G_a and
the second element is the correction coefficient γ_c that
corresponds to the algebraic codebook gain.
Accordingly, in the G.729A method, an interframe
5   prediction table is used in quantization of algebraic
codebook gain and the prediction method is the same as
that of the AMR method.

In the fourth embodiment, the sound-source signal
on the AMR side is found using dequantized values
10   obtained by the dequantizers 82a ~ 85a$_2$ from the codes
I_LAG1(m,0), I_CODE1(m,0), I_GAIN1a(m,0), I_GAIN1c(m,0)
of the AMR method and the signal is adopted as a sound-
source signal for reference purposes.

Next, pitch lag is found from the pitch-lag code
15   I_LAG2(n,0) already converted to the G.729A method and
the adaptive codebook output corresponding to this pitch
lag is obtained. Further, the algebraic codebook output
is created from the converted algebraic code
I_CODE2(n,0). Thereafter, table values are extracted
20   one set at a time in the order of the indices from the
gain quantization table for G.729A and the adaptive
codebook gain G_a and algebraic codebook gain G_c are
found. Next, the sound-source signal (sound-source
signal for testing) that prevailed when the conversion
25   was made to the G.729A method is created from the
adaptive codebook output, algebraic codebook output,
adaptive codebook gain and algebraic codebook gain, and
the error power between the sound-source signal for

reference and the sound-source signal for testing is calculated. Similar processing is executed with regard to the gain quantization table values indicated by all of the indices and the index for which the smallest

5 value of error power is obtained is adopted as the optimum gain quantization code.

The details of the processing procedure will now be described.

(1) First, adaptive codebook output $pitch_1(i)$

10 ($i = 0, 1, ..., 39$) corresponding to pitch-lag code I_LAG1 in the AMR method is found.

(2) The sound-source signal for reference is found from the following equation:

$$ex_1(i) = G_a \cdot pitch_1(i) + G_c \cdot code(i) \quad (i = 0,1,... 39)$$

15 (3) Adaptive codebook output $pitch_2(i)$ ($i = 0, 1, ..., 39$) corresponding to pitch-lag code I_LAG2(n,k) in the G.729A method is found.

(4) Table values $G_{a2}(L)$, $\gamma_c(L)$ corresponding to the Lth gain code are extracted from the gain quantization

20 table.

(5) An energy component g_c' predicted from the algebraic codebook gain of a past subframe is calculated and $G_{c2}(L) = g_c'\gamma_c(L)$ is obtained.

(6) The sound-source signal for testing is found

25 from the following equation:

$$ex_2(i,L) = G_{a2}(L) \cdot pitch_2(i) + G_{c2}(L) \cdot code(i)$$

$$(i = 0,1,...,39)$$

It should be noted that the algebraic codebook output

code(i) is the same in both the AMR and G.729A methods.

(7) The square of the error is found from the following equation:

$$E(L) = [ex_1(i) - ex_2(i,L)]^2 \qquad (i = 0,1,\ldots,39)$$

5   (8) The value of $E(L)$ is calculated with regard to the patterns ($L = 0 \sim 127$) of all indices of the gain quantization table and the L for which $E(L)$ is minimized is output as the optimum gain code I_GAIN2(n,0).

In the foregoing description, the square of the error of the sound-source signal is used as a reference when the optimum gain code is retrieved. However, an arrangement may be adopted in which reconstructed voice is found from the sound-source signal and the gain code is retrieved in the region of the reconstructed voice.

15   (e) Code transmission processing

Since the frame lengths in the ARM and G.729A methods differ from each other, two frames of channel data in the G.729A method are obtained from one frame of channel data in the AMR method. Accordingly, the buffer

20   87 (Fig. 16) inputs the codes I_LSP2(n), I_LAG2(n,0), I_LAG2(n,1), I_CODE2(n,0), I_CODE2(n,1), I_GAIN2(n,0), I_GAIN2(n,1) to the code multiplexer 86. The latter multiplexes these input codes to create the voice signal of the nth frame in the G.729A method and sends the code

25   to the transmission path as the line data.

Next, the buffer 87 inputs the codes I_LSP2(n+1), I_LAG2(n+1,0), I_LAG2(n+1,1), I_CODE2(n+1,0), I_CODE2(n+1,1), I_GAIN2(n+1,0), I_GAIN2(n+1,1) to the

code multiplexer 86. The latter multiplexes these input codes to create the voice signal of the (n+1)th frame in the G.729A method and sends the code to the transmission path as the line data.

5   (F) Fifth embodiment

The foregoing embodiments deal with cases in which there is no transmission-path error. In actuality, however, if wireless communication is employed as when using a cellular telephone, bit error or burst error

10   occurs owing to the influence of phenomena such as phasing, the voice code changes to one different from the original and there are instances where the voice code of an entire frame is lost. If traffic is heavy over the Internet, transmission delay grows, the voice

15   code of an entire frame may be lost or frames may change places in terms of their order.

(a) Effects of transmission-path error

Fig. 20 is a diagram useful in describing the effects of transmission-path error. Components in Fig.

20   20 that are identical with those shown in Figs. 1 and 2 are designated by like reference characters. This arrangement differs in that it possesses a combiner 95 which simulates the addition of transmission-path error (channel error) to a transmit signal.

25   Input voice enters the encoder 61a of encoding method 1 and the encoder 61a outputs a voice code V1 of encoding method 1. The voice code V1 enters the voice code conversion unit 80 through the transmission path

(Internet, etc.) 71, which is wired.   If channel error

intrudes before the voice code V1 enters the voice code

conversion unit 80, however, the voice code V1 is

distorted into a voice code V1', which differs from the

5   voice code V1, owing to the effects of such channel

error.   The voice code V1' enters the code separator 81,

where it is separated into the parameter codes, namely

the LSP code, pitch-lag code, algebraic code and gain

code.   The parameter codes are converted by respective

10   ones of the code converters 82, 83, 84 and 85 to codes

suited to the encoding method 2.   The codes obtained by

the conversions are multiplexed by the code multiplexer

86, whence a voice code V2 compliant with encoding

method 2 is finally output.

15   Thus, if channel error intrudes prior to input of

the voice code V1 to voice code conversion unit 80,

conversion is carried out based upon the erroneous voice

code V1' and, as a consequence, the voice code V2

obtained by the conversion is not necessarily the

20   optimum code.   With CELP, furthermore, an IIR filter is

used as a voice synthesis filter.   If the LSP code or

gain code, etc., is not the optimum code owing to the

effects of channel error, therefore, the filter often

oscillates and produces a large abnormal sound.   Another

25   problem is that because of the properties of an IIR

filter, once the filter oscillates, the vibration

affects the ensuing frame.   Consequently, it is

necessary to reduce the influence which channel error

has on the voice code conversion components.

(b)   Principles of the fifth embodiment

Fig. 21 illustrates the principles of the fifth embodiment.   Here encoding methods based upon CELP

5   compliant with AMR and G.729A are used as the encoding methods 1 and 2.

In Fig. 21, input voice xin is input to the encoder 61a of encoding method 1 so that a voice code sp1 of the encoding method 1 is produced.   The voice code sp1 is

10   input to the voice code conversion unit 80 through a wireless (radio) channel or wired channel (the Internet).   If channel error ERR intrudes before the voice code sp1 enters the voice code conversion unit 80, the voice code sp1 is distorted into a voice code sp'

15   that contains the channel error.   The pattern of the channel error ERR depends upon the system and various patterns are possible, examples of which are random-bit error and burst error.   If no error intrudes upon the input, then the codes sp1' and sp1 will be identical.

20   The voice code sp1' enters the code separator 81 and is separated into LSP code LSP1, pitch-lag code Lag1, algebraic code PCB1 and gain code Gain1.   The voice code sp1' further enters a channel-error detector 96, which detects through a well-known method whether

25   channel error is present or not.   For example, channel error can be detected by adding a CRC code onto the voice code sp1 in advance or by adding data, which is indicative of the frame sequence, onto the voice code

sp1 in advance.

The LSP code LSP1 enters an LSP correction unit 82c, which converts the LSP code LSP1 to an LSP code LSP1' in which the effects of channel error have been reduced. The pitch-lag code Lag1 enters a pitch-lag correction unit 83c, which converts the pitch-lag code Lag1 to a pitch-lag code Lag1' in which the effects of channel error have been reduced. The algebraic code PCB1 enters an algebraic-code correction unit 84c, which converts the algebraic code PCB1 to an algebraic code PCB1' in which the effects of channel error have been reduced. The gain code Gain1 enters a gain-code correction unit 85c, which converts the gain code Gain1 to a gain code Gain1' in which the effects of channel error have been reduced.

Next, the LSP code LSP1' is input to the LSP code converter 82 and is converted thereby to an LSP code LSP2 of encoding method 2, the pitch-lag code Lag1' is input to the pitch-lag code converter 83 and is converted thereby to an pitch-lag code Lag2 of encoding method 2, the algebraic code PCB1' is input to the algebraic code converter 84 and is converted thereby to an algebraic code PCB2 of encoding method 2, and the gain code Gain1' is input to the gain code converter 85 and is converted thereby to a gain code Gain2 of encoding method 2.

The codes LSP2, Lag2, PCB2 and Gain2 are multiplexed by the code multiplexer 86, which outputs a

voice code sp2 of encoding method 2.

By adopting this arrangement, it is possible to diminish a decline in post-conversion voice quality due to channel error, which is a problem with the conventional voice code converter.

(c)   Voice code converter according to the fifth embodiment

Fig. 22 is a block diagram illustrating the structure of the voice code converter of the fifth embodiment. This illustrates a case where G.729A and AMR are used as the encoding methods 1 and 2, respectively. It should be noted that although there are eight AMR encoding modes, Fig. 22 illustrates a case where 7.94 kbps is used. In Fig. 22, a voice code $sp1(n)$, which is a G.729A-compliant encoder output of an nth frame, is input to the voice code conversion unit 80. Since the G.729A bit rate is 8 kbps, $sp1(n)$ is represented by a bit sequence of 80 bits. The code separator 81 separates the voice code $sp1(n)$ into the LSP code $LSP1(n)$, pitch-lag code $Lag1(n,j)$, algebraic code $PCB1(n,j)$ and gain code $Gain1(n,j)$. The suffix j in the parentheses represents the subframe number and takes on values of 0 and 1.

If channel error ERR intrudes before the voice code $sp1(n)$ enters the voice code conversion unit 80, the voice code $sp1(n)$ is distorted into a voice code $sp1'(n)$ that contains the channel error. The pattern of the channel error ERR depends upon the system and various

patterns are possible, examples of which are random-bit error and burst error. If burst error occurs, the information of an entire frame is lost and voice cannot be reconstructed correctly. Further, if voice code of a

5   certain frame does not arrive within a prescribed period of time owing to network congestion, this situation is dealt with by assuming that there is no frame. As a consequence, the information of an entire frame may be lost and voice cannot be reconstructed correctly. This

10   is referred to as "frame disappearance" and necessitates measures just as channel error does. If no error intrudes upon the input, then the codes sp1'(n) and sp1(n) will be exactly the same.

The particular method of determining whether

15   channel error or frame disappearance has occurred or not differs depending upon the system. In the case of a cellular telephone system, for example, the usual practice is to add an error detection code or error correction code onto the voice code. The channel-error

20   detector 96 is capable of detecting whether the voice code of the present frame contains an error based upon the error detection code. Further, if the entirety of one frame's worth of voice code cannot be received within a prescribed period of time, this frame can be

25   dealt with by assuming frame disappearance.

The LSP code LSP1(n) enters the LSP correction unit 82c, which converts this code to an LSP parameter lsp(i) in which the effects of channel error have been reduced.

The pitch-lag code Lag1(n,j) enters the pitch-lag correction unit 83c, which converts this code to a pitch-lag code Lag1'(n,j) in which the effects of channel error have been reduced. The algebraic code

5   PCB1(n,j) enters the algebraic-code correction unit 84c, which converts this code to an algebraic code PCB1'(n,j) in which the effects of channel error have been reduced. The gain code Gain1(n,j) enters the algebraic-code correction unit 85c, which converts this code to a pitch

10 gain Ga(n,j) and algebraic codebook gain Gc(n,j) in which the effects of channel error have been reduced.

If channel error or frame disappearance has not occurred, the LSP correction unit 82c outputs an LSP parameter lsp(i) that is identical with that of the

15 first embodiment, the pitch-lag correction unit 83c outputs a code, which is exactly the same as Lag1(n,j), as Lag1'(n,j), the algebraic-code correction unit 84c outputs a code, which is exactly the same as PCB1(n,j), as PCB1'(n,j), and the gain-code correction unit 85c

20 outputs a pitch gain Ga(n,j) and algebraic codebook gain Gc(n,j) that are identical with those of the first embodiment.

(d)   LSP code correction and LSP code conversion

The LSP correction unit 82c will now be described.

25 If an error-free LSP code LSP1(n) enters the LSP correction unit 82c, the latter executes processing similar to that of the LSP dequantizer 82a of the first embodiment. That is, the LSP correction unit 82c

divides LSP1(n) into four smaller codes L_0, L_1, L_2 and L_3.
The code L_1 represents an element number of the LSP
codebook CB1, and the codes L_2, L_3 represent element
numbers of the LSP codebooks CB_2, CB_3, respectively. The

5   LSP codebook CB1 has 128 sets of 10-dimensional vectors,
and the LSP codebooks CB2 and CB3 both have 32 sets of
5-dimensional vectors. The code L_0 indicates which of
two types of MA prediction coefficients (described
later) to use. A residual vector $l_i^{(n)}$ of the nth frame

10 is found by the following equation:

$$l_i^{(n)} = \begin{cases} CB1(L_1, i) + CB2(L_2, i) & (i = 1, \ldots, 5) \\ CB1(L_1, i) + CB3(L_3, i-5) & (i = 6, \ldots, 10) \end{cases} \qquad (38)$$

Next, an LSF coefficient $\omega(i)$ is found from the
residual vector $l_i^{(n)}$ and residual vectors $l_i^{(n-k)}$ of the
last most recent four frames in accordance with the

15 following equation:

$$\omega(i) = (1 - \sum_{k=1}^{4} p(i, k)) l_i^{(n)} + \sum_{k=1}^{4} p(i, k) l_i^{(n-k)} \qquad , (i = 1, \ldots, 10) \quad (39)$$

where p(i,k) represents which coefficient of the two
types of MA prediction coefficients has been specified
by the code L_0. The residual vector $l_i^{(n)}$ is held in a

20 buffer 82d for the sake of frames from the next frame
onward. Thereafter, the LSP correction unit 82c finds
the LSP parameter lsp(i) from the LSP coefficient $\omega(i)$
using the following equation:

$$lsp(i) = \cos[\omega(i)] \qquad (i = 1, \cdots, 10) \qquad (40)$$

25   Thus, if channel error or frame disappearance has
not occurred, the input to the LSP code converter 82 can
be created by calculating LSP parameters, through the
above-described method, from LSP code received in the

present frame and LSP code received in the past four
frames.

The above-described procedure cannot be used if the
correct LSP code of the present frame cannot be received
owing to channel error or frame disappearance. In the
fifth embodiment, therefore, if channel error or frame
disappearance has occurred, the LSP correction unit 82c
uses the following Equation to create the residual
vector $l_i^{(n)}$ from the past four good frames of LSP code
received last:

$$l_i^{(n)} = \left[\hat{\omega}_i^{(m)} - \sum_{k=1}^{4} p(i,k) l_i^{(n-k)}\right] / \left[1 - \sum_{k=1}^{4} p(i,k)\right], \quad (i = 1,\ldots,10) \quad (41)$$

where p(i,k) represents the MA prediction coefficient of
the last good frame received.

Thus, as set forth above, the residual vector $l_i^{(n)}$
of the present frame can be found in accordance with
Equation (41) in this embodiment even if the voice code
of the present frame cannot be received owing to channel
error or frame disappearance.

The LSP code converter 82 executes processing
similar to that of the LSP quantizer 82b of the first
embodiment. That is, the LSP parameter lsp(i) from the
LSP correction unit 82c is input to the LSP code
converter 82, which then proceeds to obtain the LSP code
for AMR by executing dequantization processing identical
with that of the first embodiment.

  (e) Pitch-lag correction and pitch-lag code
    conversion

The pitch-lag correction unit 83c will now be

described. If channel error and frame disappearance have not occurred, the pitch-lag correction unit 83c outputs the received lag code of the present frame as Lag1'(n,j). If channel error or frame disappearance has

5 occurred, the pitch-lag correction unit 83c acts so as to output, as Lag1'(n,j), the last good frame of pitch-lag code received. This code has been stored in buffer 83d. It is known that pitch lag generally varies gradually in voiced segments. In a voiced segment,

10 therefore, there is almost no decline in sound quality even if the pitch lag of the preceding frame is substituted, as mentioned earlier. It is known that pitch lag undergoes a large conversion in unvoiced segments. However, since the contribution in the

15 adaptive codebook in unvoiced segments is small (pitch gain is low), there is almost no decline in sound quality caused by the above-described method.

The pitch-lag code converter 83 performs the same pitch-lag code conversion as that of the first

20 embodiment. Specifically, whereas frame length according to the G.729A method is 10 ms, frame length according to AMR is 20 ms. When pitch-lag code is converted, therefore, it is necessary that two frame's worth of pitch-lag code according to G.729A be converted

25 as one frame's worth of pitch-lag code according to AMR. Consider a case where pitch-lag codes of the nth and (n+1)th frames in the G.729A method are converted to pitch-lag code of the mth frame in the AMR method. A

pitch-lag code is the result of combining integral lag and non-integral into one word. In even-numbered subframes, the methods of synthesizing pitch-lag codes in the G.729A and AMR methods are exactly the same and the numbers of quantization bits are the same, i.e., eight. This means that the pitch-lag code can be converted in the manner indicated by the following equations:

$$LAG2(m,0) = LAG1'(n,0) \qquad (42)$$

$$LAG2(m,2) = LAG1'(n+1,0) \qquad (43)$$

Further, quantization of the difference between integral lag of the present frame and integral lag of the preceding subframe is performed in common for the odd-numbered subframes. However, since the number of quantization bits is one larger for the AMR method, the conversion can be made as indicated by the following equations:

$$LAG2(m,1) = LAG1'(n,1) + 15 \qquad (44)$$

$$LAG2(m,3) = LAG1'(n+1,1) + 15 \qquad (45)$$

(f) Algebraic code correction and algebraic code conversion

If channel error and frame disappearance have not occurred, the algebraic-code correction unit 84c outputs the received algebraic code of the present frame as $PCB1'(n,j)$. If channel error or frame disappearance has occurred, the algebraic-code correction unit 84c acts so as to output, as $PCB1'(n,j)$, the last good frame of algebraic code received. This code has been stored in

buffer 84d.

The algebraic code converter 84 performs the same algebraic code conversion as that of the first embodiment. Specifically, although frame length in the G.729A method differs from that in the AMR method, subframe length is the same for both, namely 5 ms (40 samples). Further, the structure of the algebraic code is exactly the same in both methods. Accordingly, the pulse positions and the pulse polarity information that are the results output from the algebraic codebook search in the G.729A method can be replaced as is on a one-to-one basis by the results output from the algebraic codebook search in the AMR method. The algebraic-code conversions are as indicated by the following:

$$PCB2(m,0) = PCB1'(n,0) \qquad (46)$$

$$PCB2(m,1) = PCB1'(n,1) \qquad (47)$$

$$PCB2(m,2) = PCB1'(n+1,0) \qquad (48)$$

$$PCB2(m,3) = PCB1'(n+1,1) \qquad (49)$$

(g) Gain code correction and gain code conversion

If channel error and frame disappearance have not occurred, the gain-code correction unit 85c finds the pitch gain $Ga(n,j)$ and the algebraic codebook gain $Gc(n,j)$ from the received gain code $Gain1(n,j)$ of the present frame in a manner similar to that of the first embodiment. However, in accordance with the G.729A method, the algebraic codebook gain is not quantized as

is. Rather, quantization is performed with the participation of the pitch gain Ga(n,j) and a correction coefficient γ_c for algebraic codebook gain.

Accordingly, when the gain code Gain1(n,j) is input
5   thereto, the gain-code correction unit 85c obtains the pitch gain Ga(n,j) and correction coefficient γ_c corresponding to the gain code Gain1(n,j) from the G.729A gain quantization table. Next, using the correction coefficient γ_c and prediction value g_c', which
10   is predicted from the logarithmic energy of algebraic codebook gain of the past four subframes, the gain-code correction unit 85c finds algebraic codebook gain Gc(n,j) in accordance with Equation (21).

If channel error or frame disappearance has
15   occurred, the gain code of the present frame cannot be used. Accordingly, pitch gain Ga(n,j) and algebraic codebook gain Gc(n,j) are found by attenuating the gain of the immediately preceding subframe stored in buffers 85d1, 85d2, as indicated by Equations (50) to (53)
20   below. Here α, β are constants equal to 1 or less. Pitch gain Ga(n,j) and algebraic codebook gain Gc(n,j) are the outputs of the gain-code correction unit 85c.

$$Ga(n,0) = \alpha \cdot Ga(n-1) \qquad\qquad (50)$$

$$Ga(n,1) = \alpha \cdot Ga(n,0) \qquad\qquad (51)$$

25   $$Gc(n,0) = \beta \cdot Gc(n-1,1) \qquad\qquad (52)$$

$$Gc(n,1) = \beta \cdot Gc(n,0) \qquad\qquad (53)$$

Gain converters 85b$_1$', 85b$_2$' will now be described.

In the AMR method, pitch gain and algebraic

codebook gain are quantized separately. However,
algebraic codebook gain is not quantized directly.
Rather, a correction coefficient for algebraic codebook
gain is quantized. First, pitch gain Ga(n,0) is input

5   to pitch gain converter $85b_1$' and is subjected to scalar
quantization. Values of 16 types (four bits) the same
as those of the AMR method have been stored in the
scalar quantization table. The quantization method
includes calculating the square of the error between the

10 pitch gain Ga(n,0) and each table value, adopting the
table value for which the smallest error is obtained as
the optimum value and adopting this index as the
gain2a(m,0).

The algebraic codebook gain converter $85b_2$' scalar-

15 quantizes $\gamma_c(n,0)$. Values of 32 types (five bits) the
same as those of the AMR method have been stored in this
scalar quantization table. The quantization method
includes calculating the square of the error between
$\gamma_c(n,0)$ and each table value, adopting the table value

20 for which the smallest error is obtained as the optimum
value and adopting this index as the gain2c(m,0).

Similar processing is executed to find Gain2a(m,1)
and Gain2c(m,1) from Gain1(n,1). Further, Gain2a(m,2)
and Gain2c(m,2) are found from Gain1(n+1,0), and

25 Gain2a(m,3) and Gain2c(m,3) are found from Gain1(n+1,1).

(h)   Code multiplexing

The code multiplexer 86 retains converted code
until the processing of two frame's worth (one frame's

worth in the AMR method) of G.729A code is completed,
processes two frames of the G.729A code and outputs
voice code sp2(m) when one frame's worth of AMR code has
been prepared in its entirety.

5   Thus, as described above, this embodiment is such
that if channel error or frame disappearance occurs, it
is possible to diminish the effects of the error when
G.729A voice code is converted to AMR code. As a
result, it is possible to achieve excellent voice
10   quality in which a decline in the quality of sound is
diminished in comparison with the conventional voice
code converter.

Thus, in accordance with the present invention,
codes of a plurality of components necessary to
15   reconstruct a voice signal are separated from a voice
code based upon a first voice encoding method, the code
of each component is dequantized and the dequantized
values are quantized by a second encoding method to
achieve the code conversion. As a result, delay can be
20   reduced over that encountered with the conventional
tandem connection and a decline in sound quality can be
reduced as well.

Further, in accordance with the present invention,
in a case where LSP code of the first excitation signal
25   is dequantized and a dequantized value LSP1(i) is
quantized by the second encoding method to achieve the
code conversion when a conversion of LSP code is
performed, not only a first distance (error) between the

dequantized value LSP1(i) and a dequantized value LSPc3(i) of LSP code obtained by conversion but also a second distance (error) between an intermediate LSP code dequantized value LSP0(i) of the first encoding method

5 and an intermediate LSP code dequantized value LSPc1(i) of the second encoding method calculated by interpolation is taken into account and input to achieve the LSP code conversion. As a result, it is possible to perform an excellent voice code conversion with little

10 conversion error even in a case where the quality of input voice varies within the frame.

Further, in accordance with the present invention, the first and second distances are weighted and an LPC coefficient dequantized value LSP1(i) is encoded to an

15 LPC code in the second encoding method in such a manner that the sum of the weighted first and second distances will be minimized. This makes it possible to perform a voice code conversion with a smaller conversion error.

Further, in accordance with the present invention,

20 LPC coefficients are expressed by n-order vectors, the n-order vectors are divided into a plurality of small vectors (low-, midrange- and high-frequency vectors), a plurality of code candidates for which the sum of the first and second distances will be small is calculated

25 for each small vector, codes are selected one at a time from the plurality of code candidates of each small vector and are adopted as n-order LPC codes, and an n-order LPC code is decided based upon a combination for

which the sum of the first and second distances is minimized. As a result, a voice code conversion that makes possible the reconstruction of sound of higher quality can be performed.

5   Further, in accordance with the present invention, it is possible to provide excellent reconstructed voice after conversion by diminishing a decline in sound quality caused by channel error, which is a problem with the conventional voice code converter. In particular,

10   in the case of CELP algorithms used widely in low-bit-rate voice encoding in recent years, an IIR filter is used as a voice synthesis filter and, as a result, the system is susceptible to the influence of channel error and large abnormal sounds are often produced by

15   oscillation. The improvement afforded by the present invention is especially effective in dealing with this problem.

It should be noted that although the present invention has been described with regard to voice

20   signals and voice codes, it is applicable to other sound-related signals and codes, which may be referred to as "acoustic signals" and "acoustic codes".

As many apparently widely different embodiments of the present invention can be made without departing from

25   the spirit and scope thereof, it is to be understood that the invention is not limited to the specific embodiments thereof except as defined in the appended claims.